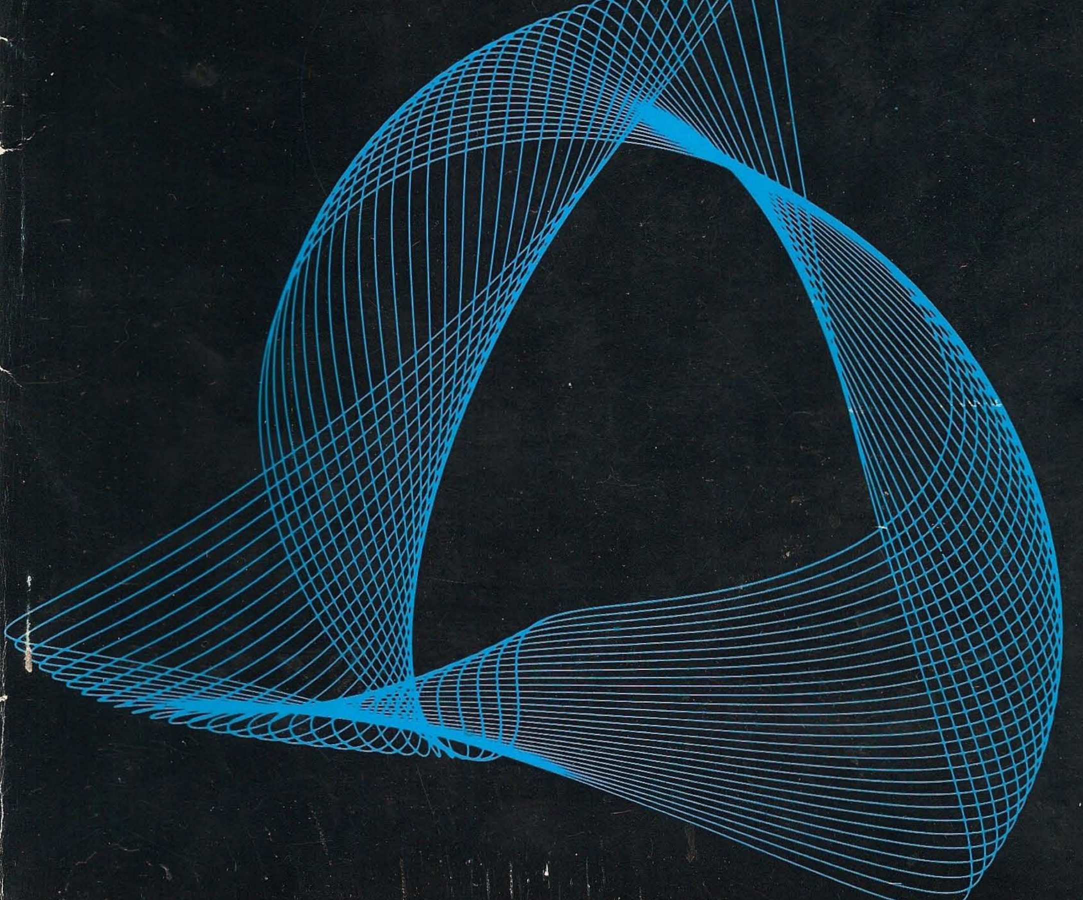
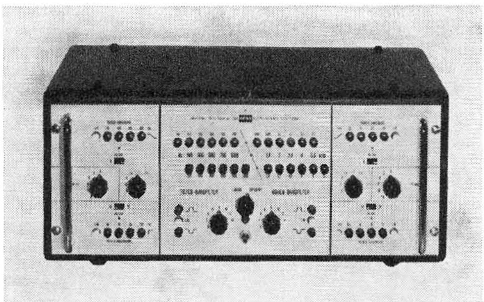
Electronic Music Review



UNIVERSAL EQUALIZER UE 100



Modern techniques of recording, broadcasting, film, and television have made equalizers indispensable to the control of tone quality as well as an integral part of creative expression.

The UE 100 is capable of reproducing known frequency response curves, compensating for frequency-dependent losses in transmission, and adjusting playback tone-color to the individual desires of the artist and sound engineer.

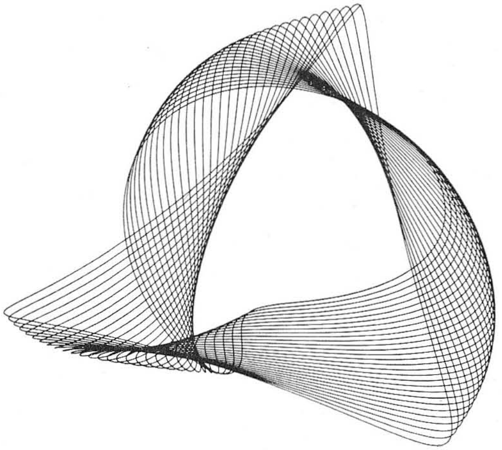
KLEIN + HUMMEL

7301 Kemnat Germany
Cable Telewatt Stuttgart
Phone Stuttgart 25 32 46

Representatives:	Bruxelles	Electronique Generale, 14 Rue Pere de Deken
	Paris	A. Frei, 172 Rue de Courcelles
	New York	Gotham Audio Corp., 2 W. 46 St.
	Tokyo	Kawamura Lab., 34 Yarai-Cho

Electronic Music Review

No. 1 January 1967



Reynold Weidenaar, Editor
Robert A. Moog, Technical Editor

Electronic Music Review, No. 1, January 1967. Published quarterly by the Independent Electronic Music Center, Inc., Trumansburg, N. Y. 14886.

Personal subscriptions available through IEMC membership (annual dues $6). Institutional subscriptions, one year $8, two years $15. Outside North America, add 50c per year, payable in U.S. funds.

© Independent Electronic Music Center, Inc. 1967

Contents

EMscope

Electronic Music Review has been established to provide a source of information and a means of discourse on all aspects of electronic music. We hope to provide some relief from several serious problems facing those interested in this medium: first, the difficulties in communication between scientists and musicians; second, the lack of practical advice composers need in establishing and using their studios; third, the considerable research required to gather even a minimal amount of useful information on the subject.

To meet these needs, we will provide a complete spectrum of technical information, from simple introductory and how-to-build-it features to articles describing recent developments in sophisticated electronic music technology. These advanced articles will be prefaced by special introductions defining the terms and concepts unfamiliar to many musicians. In addition, we invite readers to submit technical questions of practical interest to be answered in the pages of EMR (no personal replies—address Technical Editor, Electronic Music Review, Trumansburg, N.Y. 14886). A comprehensive discussion of mixers will appear in the July issue and a review of tape recorders suitable for electronic music will appear in the October issue. Both of these articles will contain an introduction to operating principles, practical pointers on using the devices, and a listing of commercially available instruments. We invite potential contributors to submit articles on the design or use of mixers or tape recorders, or on any other topic of interest to EMR readers. Furthermore, various reference information and announcements appear in this and subsequent issues. The April issue of EMR will consist of the most extensive catalog of electronic music compositions that has ever appeared (see special announcement elsewhere in this issue).

The Cover

Two sine waves an octave apart and a device invented by physicist Edward Lias resulted in the cover drawing, called a "cosmograph."

In Memoriam

Hermann Heiss, director of the Darmstadt electronic music studio, died on December 6, 1966 in Darmstadt.

Heiss was born in Darmstadt on December 29, 1897; he studied counterpoint with Sekles and piano with Renner and Hoehn, but was primarily self-taught. Heiss was acquainted with Schoenberg and with Hauer, whose *Zwoelftontechnik* was dedicated to Heiss. Hauer's influence was instrumental in Heiss' transition to twelve-tone music in 1923. Heiss' compositional techniques are described in his *Tonbewegungslehre* (1949). From 1946 Heiss taught at the International Summer Courses for New Music in Darmstadt, and from 1953 he headed the advanced composition course at the Staedtischen Akademie fuer Tonkunst in Darmstadt.

Always ahead of his time, Heiss was one of the first composers of electronic music, and in 1957 he took advantage of the opportunity to design and

establish the Darmstadt electronic music studio.

Although most of his works were burned during the war, several of Heiss' most important compositions survive:
(Instrumental) *Capricci Ritmici fuer Klavier; Sinfonia Athematica; Konfigurationen fuer Orchester.*
(Electronic) *Zuordnung Vier; Missa* (with choir and soloists); *Variable Musik.*
—Klaus Dienert, Assistant to Hermann Heiss

Darmstadt

This year's International Summer Courses for New Music in Darmstadt will include lectures on "Problems of Electronic Sound Material" by Jozef Patkowski, director of the electronic music studio at Polskie Radio, Warsaw. Further information available from Internationale Musikinstitut, 61 Darmstadt, Nieder-Ramstaedterstr. 190, Germany.

Summer Workshop

A five-day workshop entitled Exploratory Electronics will be offered July 24-28 at Peabody Conservatory in Baltimore. Aimed at introducing electronic music to school music teachers and others, the workshop will be conducted by Jean Eichelberger Ivey (who has worked at the electronic music studios of the University of Toronto and Brandeis University). Further information available from Ray E. Robinson, Director of Summer Session, Peabody Conservatory, Baltimore, Maryland 21202.

Recent Publications

Adorno, et al. *Form in der Neuen Musik.* 1966. Schott, Mainz, Germany.

Barbaud, Pierre. *L'initiation a la Composition Musicale Automatique.* 1965. Dunod, 92, Rue Bonaparte, Paris 6e, France.

Bibliographie Concernant la Recherche Musicale. 1966. Groupe de Recherches Musicales, Centre Bourdan, 5, Av. due Recteur Poincare, Paris 16e, France.

Gaudeamus (contemporary music newsletter—first issue). March 1967. Gaudeamus, Postbox 30, Bilthoven, Netherlands.

Henry, Otto. *A Preliminary Checklist: Books and Articles on Electronic Music.* 1966. Otto Henry, 2114 Milan, New Orleans, Louisiana 70115.

Hiller, Lejaren. *Informationstheorie und Computermusik.* 1964. Schott, Mainz, Germany.

Nutida Musik (ISCM 1966 issue). Sveriges Radio, Box 955, Stockholm 1, Sweden.

Source—Music of the Avant Garde (first issue). January 1967. Composer/Performer Edition, 330 University Ave., Davis, California 95616.

Recent Records

Helidor HS25047—1967—"Electronic Music from the University of Illinois"—Herbert Bruen, Kenneth Gaburo, Charles Hamm, Lejaren Hiller, Salvatore Martirano.

Owl ORLP6,7,8—1966—"Organized Sound by Tod Dockstader" (review by Kurt Stone to appear in the July EMR).

Sveriges Radio (Box 955, Stockholm 1, Sweden) LPD1—1966—"Dokumentation fran Elektronmusikstudion"—Ralph Lundsten, Leo Nilson.

Forthcoming Concerts

Cleveland—June 7, Kulas Hall; Donald Erb.

New York—March 27, Town Hall; Jacob Druckman. April 24, Town Hall; Larry Austin, Raymond Wilding-White. May 15, Town Hall; Henry Brant, Donald Erb.

Please Note

Information on recent records and publications, forthcoming concerts, lectures, seminars, etc., should reach EMR no later than one month before month of publication.

KEEP INFORMED!

—on national trends in the musical scene

—on proposed Congressional legislation affecting music

—on the activities of 57 NMC member organizations with a combined membership of over 1,250,000

—on Government activities in the field of music

—on contests and competitions here and abroad

—on music activities in UNESCO

SUBSCRIBE TO THE
NMC BULLETIN

a 28-36 page magazine published without advertising

$3.50 yearly $1.50 per copy

NATIONAL MUSIC COUNCIL
Dept. E
2109 Broadway
New York, N. Y. 10023

Happy Birthday

Raymond Wilding-White

A lot of tape has run through the capstan in the near two decades since Pierre Schaeffer, as he expressed it, abandoned the recording studio and took refuge in the control booth. At that time, concrete and electronic music were infants sharing the nursery with the jet, the computer, and the Bomb. A new hi-fi millenium was at hand that would bring an end to the 9-KC cutoff and bring Mozart to everyone; and a new and exciting musical concept was at hand—the composer would have for a palette "all sounds," no less. But at 18 a boy comes to a man's estate and in the coming decade this family of the forties, the new music included, will not be judged by its juvenile potential, but by its adult results.

With scores of studios, hundreds of composers, and thousands of works, the future of the tape deck as a technique is assured; but these elements alone do not ensure good health and a long life. A broader base is needed for that, and this base, like it or not, has been brought about more by the efforts of the Sony and Heath companies, the DJ and the Beatles, and the ad and A&R man than by Paris, Cologne, and Columbia-Princeton.

True that, under the ministrations of Sony et altera, the 9-KC cutoff (or something like it) is still with us, and Mozart for everyone reads more like Muzak for everyplace. But electronic music still needs the hardware of the audio industry for its production and reproduction; the problem is not whether you can live without the mass monolith—you can't—but whether it will eat you. The man most aware of this is the new manufacturer of equipment specifically designed for electronic music (there are three at present and if all goes well there will soon be more); and his appearance is timely, his survival essential. To date, studio design has been an exhilarating free-for-all of ingenious amateurs and adaptable radio-men, but this cannot be the pattern of the future, any more than the literature of the violin, the piano, or the clarinet could progress without Stradivarius, Steinway, or Selmer. The role played by the manufacturer will depend in part on his professional integrity—whether he will maintain the standards of a McIntosh or a Holtkamp—and partly on the standards set by his clientele. In this regard, please recall the dismal history of the organ from 1800 to almost today.

For standards of quality to exist there first has to be a market. Discounting the potential of the producers of commercials and the rock factories, which may well be the bread-and-butter of this new industry, academia is the obvious immediate market. Not only are new Institutions of Higher Learning sprouting out of the ground at the rate of one every week but, under the pressure of academic competition (Progress is their Principal Product), the older antagonism to the new music is being replaced by interest and even enthusiasm. The most receptive audience is the present college generation, a generation that, having grown up with an audio kit in one hand and a

slab guitar in the other, has no bias against the electronic production of music. Granted, there is a predictable time delay before this enthusiasm reaches the budget-making level (if enthusiasm ever reaches this level); still, the enthusiasm is there and is more in evidence in the younger institution than in the older: the trappers in Alaska will dance a sine-wave frug before the dons of Harvard will.

Dave Brubeck appreciated, and exploited, this vast reserve years ago, and a brand of jazz developed that was quickly labeled "college commercial." Taking a page from Brubeck, the new composer is very hip to this scene and, unlike his forebears, quite willing to hustle. Today, all God's chillun got chutzpah, even if it isn't always upper Broadway style.

The rush to the studio has started and, within ten years, there probably will not be a college with a music program that will not have a "studio." Even at that, we are talking about a limited horizon and projecting present practice into the future. But there are pitfalls to the ubiquitous college. The main one is that "them that's got, gits," and thus all the flossy hardware has gone to school, leaving the independent studio to survive on will power and a soldering gun.

In days of yore a composer could create with paper, pencil, and possibly a beat-up piano, knowing that the performance would be on worthy instruments; but not so the tape composer. Thus, if he "ain't got," he does what he can with what there is and, developing a philosophy of poverty, makes a virtue of junk. As a result, by and large, the big plants have veered towards Webern and the little guys have veered towards Cage. There is a vacuum here that can be filled by the introduction of good-quality, low-cost studio units—the *portatif* of the tape world—but, unfortunately, crucial though the need, it is the demand with least appeal for the manufacturer, since it is characterized by low cost and low profit, with a small market. Unattractive as it may be, this need has got to be filled or it will go on being filled willy-nilly by the method now in fashion.

Some would have it that the composer has become a "team man," but where his creative process is concerned, this is not true; he is still the individualist he always was. It is unlikely that he will willingly stand in line for time on "the big facility" when he can cook up his own.

Schaeffer stepped from the studio into the control booth. We are still stuck in the control booth. The main change the future must bring is the growth of a true electronic instrument that can serve for performance as well as composition. I do not mean by this the jury-rigged forest of wires we are all familiar with, or the ponderous console of a synthesizer, but a flexible modular system capable of almost endless variety.

Contrary to some early predictions, the concert hall has not gone the way of vaudeville; the live performer has refused to lie down and play dead. In fact, the loudspeaker, standing alone on stage in all its nakedness, welcomes the reassuring presence of cello players, dancers, and men with Lekolites. Though concert music as concert music, stage as stage, and dance

as dance go on—as well they should—the boundaries are no longer fixed. The tape deck is omnipresent; its growth into a full-fledged performance instrument is essential.

The same can be said of the sweeping claims for the supremacy of the computer (and its rapidly obsolescing forerunner, the synthesizer), in a way the last stand of that twenty-year-old quest for "a source of all sounds." The term "all sounds" will be meaningful only on the day the universe ends; and by then it will be too late for computers to synthesize them. Mr. Babbitt's objections to the contrary, there is validity in the statement that "if you want an oboe sound, hire an oboist." The computer can no more get along without the electronic studio than the studio can live without the traditional performer. Furthermore, if the composer can ill afford a $15,000 studio, he can less afford a $1,000,000 UNIVAC. And if three faculty composers and five students form a long line to a studio, two major companies and a gaggle of graduate students make an equally long line to the card-punch. I am by no means underselling the formidable potential of the computer, I only wish to put it in perspective.

It is rather poetic that Ferretti of M.I.T. set out sixteen years ago to create essentially a performable computer. One of the characteristics of the so-called "third generation" of computers is the appearance of the small-size specialized instrument. Ferretti's hope may be realized.

It is obvious that I have steered clear of a discussion of the kind of music that electronic music is or should be. Important though aesthetic problems may be, they are not unique criteria for evaluating future development. People seldom stop to realize that the illustrious development of the piano, which so deeply affected the course of music, was not based on the premise that its user would be either Liszt or Liberace. Present practitioners do the future of electronic music a disservice when they hold to the dogma that if it can't do everything it is beneath them, or to the opposite dogma that if you can't get it at Uncle Joe's Salvage, don't mess with it.

Electronic music has passed through its childhood and its pioneers will be dutifully praised by future generations of musicologists. Now it has taken its Bar Mitzvah. May it long flourish and have happy days.

The Multiplier-Type Ring Modulator

Harald Bode

Introduction—*Robert A. Moog*

Vibrations of the air in the frequency range of 20-20,000 cycles per second are perceived as sound. The unit of frequency is the HERTZ (Hz): 1 Hz = 1 cycle per second. The WAVELENGTH of an acoustical vibration is the distance in space spanned by one cycle, and is inversely proportional to the frequency. The WAVEFORM is a graph of the instantaneous amplitude of the vibration versus either space or time. The SINE waveform is a special waveform which consists of only one frequency. All other waveforms may be synthesized from sine waves of various frequencies, or from noise bands which can be mathematically described as continuous frequency distributions of sine waves. The SPECTRUM of a sound event is a listing of the frequencies of both the sine waves and the continuous noise bands that comprise it. The entries on this list are known as COMPONENTS. A BAND is a segment of a spectrum and is defined by its CENTER FREQUENCY, the frequency which lies at the center of the band, and its BANDWIDTH, the frequency range spanned by the band. UNIT BANDWIDTH is a band 1 Hz wide.

A VOLTAGE is a quantity of electrical force that is directly analogous to air pressure. Sound adds rapid but small variations to the average air pressure; similar variations can occur about an average voltage. The average value (over a certain time) of a voltage is known as the DIRECT, DC, or BIAS component of that voltage. The variation which that voltage undergoes about its average value is known as the SIGNAL or AC component of that voltage. When signal voltages are referred to, STANDARD VOLTAGE LEVEL is the average variation of a voltage for which most audio equipment is designed. This level is roughly 1 volt RMS. RMS stands for root mean square, the name of a mathematical procedure for determining the average of a varying voltage. A VOLTAGE-CONTROLLED device is a device whose operating characteristics may be varied by changing the magnitude of an applied voltage. Thus, a voltage-controlled oscillator is an oscillator whose output frequency depends upon the magnitude of an applied control voltage.

A MODULATOR is a device which varies the characteristics of the signal according to the nature of an applied carrier, or control voltage. Note that this use of the word is different than the usual musical use. A RING MODULATOR is such a device with two inputs and one output. In a MULTIPLIER-TYPE RING MODULATOR, the output voltage is directly proportional to the voltages at the inputs. The instantaneous amplitude of the output is therefore proportional to the product of the instantaneous amplitudes of

the two inputs. In order to produce this type of proportionality, it is necessary to use NON-LINEAR circuit elements, which are elements that change their characteristics according to the magnitude of the voltage applied to them. The PROGRAM INPUT for a ring modulator is usually audio material which is to be processed, while the CARRIER INPUT is a generated signal or a signal otherwise more precisely controlled. Basically, however, the two inputs for the ring modulator are equivalent. The output of a ring modulator will generally contain additional frequency components which are not present in either input. These are known as MODULATION PRODUCTS or SIDEBANDS. UNWANTED MODULATION PRODUCTS are those which would not be present if the accuracy of multiplication were complete.

A GATE is an electrical switching circuit which is opened or closed by the application of a bias voltage. The QUIESCENT condition of a circuit is that which exists in the absence of an applied input signal. HETERODYNING is the production of a particular sideband through the modulation of two signals.

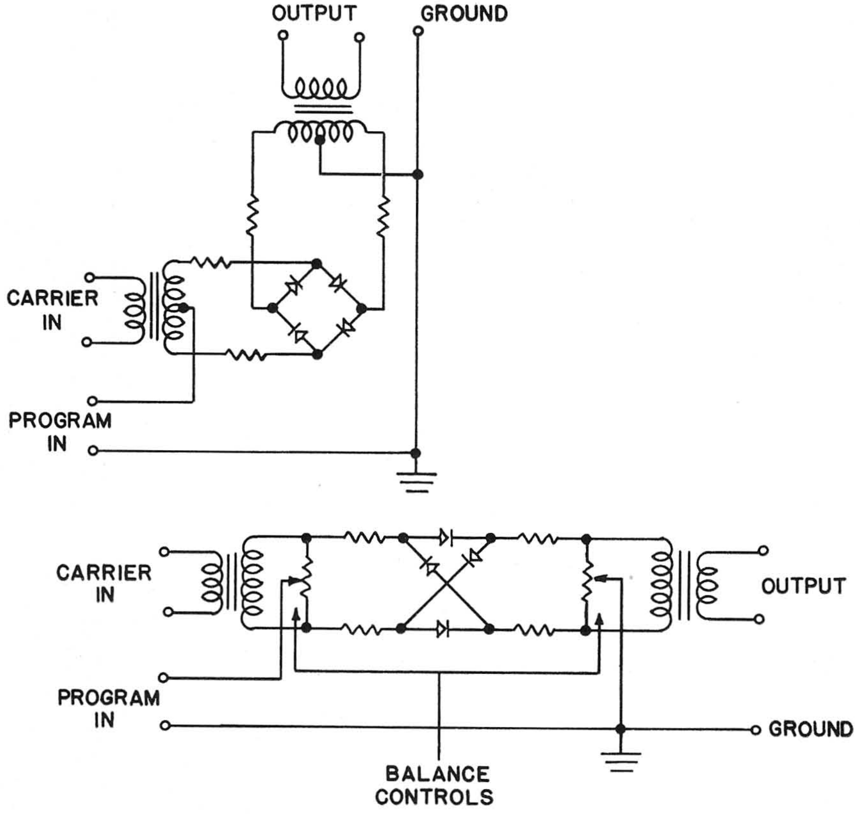


Fig. 1. (Top) Basic ring bridge circuit of a ring modulator. (Bottom) Lattice network presentation of basic ring modulator circuit with potentiometers to improve balance.

Basically two types of ring modulators are known: switching-type ring modulators and multiplier-type ring modulators. The switching-type is widely used in industrial control applications and has been discussed in some detail in the literature.[1] Because of its inherent distortions, this device is not suited for musical systems, for which the multiplier-type offers definite advantages.

All ring modulators comprise as their functional non-linear elements four diodes arranged in a ring configuration, or when redrawn, in a lattice configuration (Fig. 1). The carrier signal is introduced across two points of the bridge through the input transformer while the program input is introduced at two other points directly, one of which is grounded. The output signal is taken off two other points that are symmetrically related to the carrier input terminals. Usually the transformerless input is used for the program or control signal because of its extended frequency-handling capability from DC to very high frequencies. Its range is then limited only by the type of desired output and the limitations of the output transformer.

In industrial control applications the output waveform of the ring modulator is usually of no particular consequence, since, for instance, a servo motor does not care whether it is driven by a distorted or undistorted AC voltage. Therefore, the selection of diodes for these applications is not very critical, as long as they perform the required switching function. For this reason silicon diodes or copper oxide rectifiers are quite popular, and their choice will be tailored to the threshold level required.

A typical waveform resulting from the processing of two frequencies through a switching-type ring modulator is shown in Fig. 2. Here λ_1 is the wavelength of the lower frequency f_1 and λ_2 is the wavelength of the higher frequency f_2. The resulting sidebands are comprised of the frequencies f_2-f_1, f_2+f_1, $3f_2-f_1$, $3f_2+f_1$, $5f_2-f_1$, $5f_2+f_1$, and the further odd harmonics of f_2, minus and plus f_1. For musical purposes such a waveform is of very limited usefulness, since it sounds scratchy and unpleasant. Therefore, some people who have used this type of ring modulator in musical systems have recommended and are employing lowpass filters at the output in order to make the sound somewhat more pleasant.[2]

Problems of this type are not experienced with the multiplier-type ring modulator. This modulator uses specially selected diodes of a type which, at normal signal levels, operates in the "square law" region, and which therefore produces extremely accurate multiplication.

The output waveform resulting from the processing of two sine waves through a multiplier-type ring modulator is shown in Fig. 2.[3] The output frequencies

---

[1] Basil T. Barber, "Servo Modulators, Part One," *Control Engineering,* Aug. 1957, 65; Barber, "S.M., Part Two," *C. E.,* Oct. 1957, 96; Barber and L. S. Klivans, "S.M., Part Three," *C.E.,* Nov. 1957, 122; Klivans, "S.M., Part Four," *C.E.,* Dec. 1957, 90 (with comprehensive tables and 84 references).

[2] L. Heck and F. Buerck, "Klangumformungen in der Rundfunkstudiotechnik, insbesondere durch Anwendung der Frequenzumsetzung," *Elektronische Rundschau,* Jan. 1956.

[3] See also Harald Bode, "Sound Synthesizer Creates New Musical Effects," *Electronics,* Dec. 1, 1961.

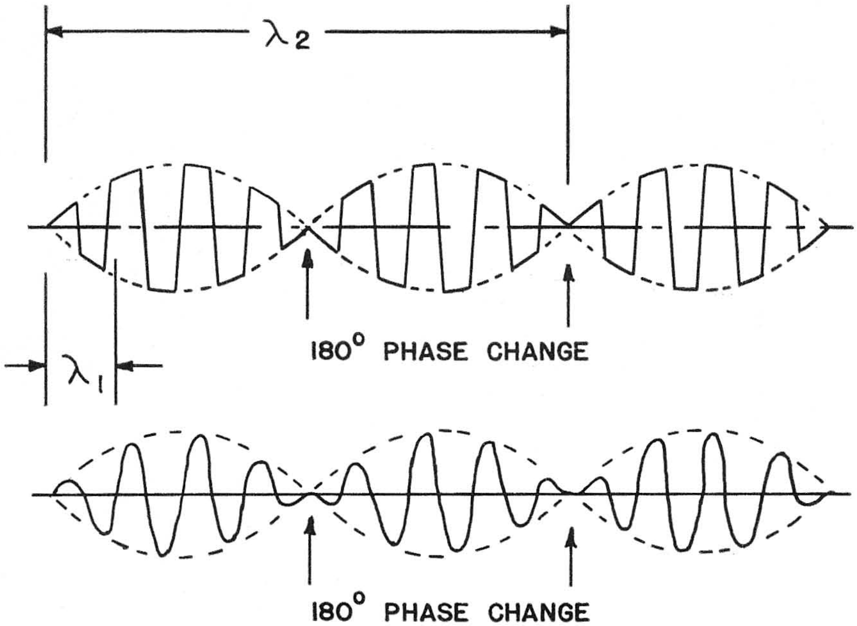


Fig. 2. (Top) Typical output waveform of switching-type ring modulator. (Bottom) Typical output waveform of multiplier-type ring modulator.

are $f_2 - f_1$ and $f_2 + f_1$ when the input frequencies are f_1 and f_2. Thus the output waveform may also be regarded as the sum of two new frequencies $f_2 - f_1$ and $f_2 + f_1$ beating together. A well designed and carefully built multiplier-type ring modulator results in an extremely small amount of unwanted modulation products.

Before discussing some applications of this device, it may be of interest to take a closer look at the electrical functions of a complete instrument that has been developed to be compatible with modern electronic music studio installations.

Basically a multiplier-type ring modulator as shown in Fig. 1 could be used as a passive, self-contained circuit or system module. However, since its output voltages are appreciably below the standard voltage levels of an electronic music system when the diodes are operating in their optimum range, an output amplifier would certainly be required. Furthermore, when the modulator is operating in the low portion of the available dynamic range, especially in pauses between events, a low level carrier feed-through may become audible. In order to prevent the carrier from being heard at all in the quiescent state, a gate is provided in the path between the carrier input terminals of the overall device and the corresponding input terminals of the ring bridge, as indicated in the block diagram of Fig. 3. This gate is activated to pass the carrier signal only when the program level exceeds a predetermined threshold voltage, which can be selected by setting the gain control

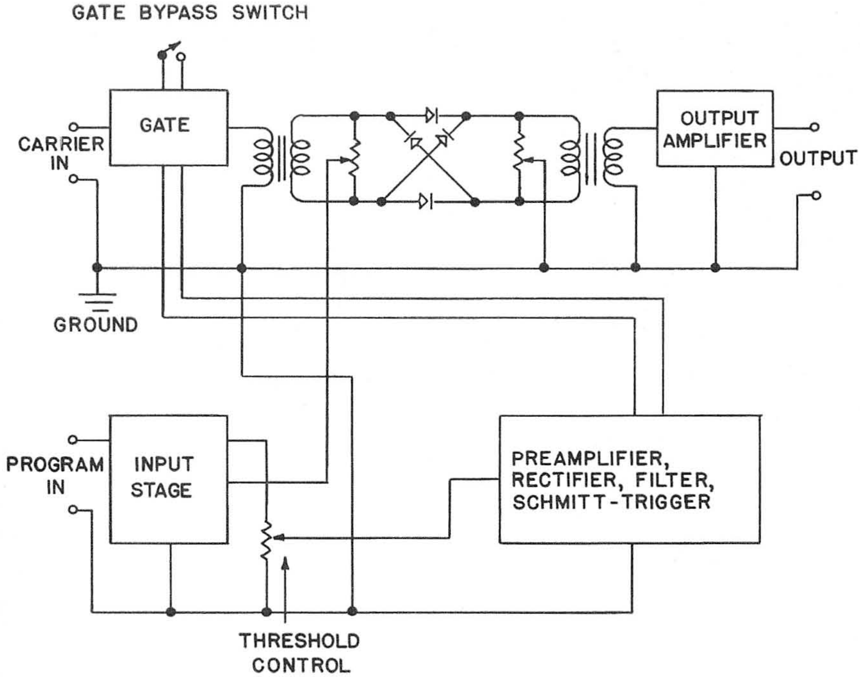


Fig. 3. Simplified block diagram of multiplier-type ring modulator with carrier input gate, threshold control preamplifier, rectifier, ripple filter, Schmitt-trigger, and output amplifier.

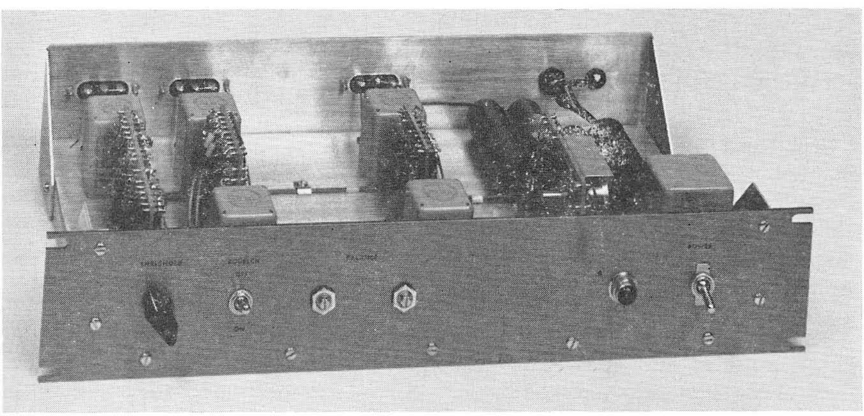


Fig. 4. Single channel Bode multiplier-type ring modulator built by the R. A. Moog Co.

of a preamplifier in the program signal path. This is followed by a rectifier, a ripple filter, and a Schmitt-trigger circuit (a circuit which has two stable states depending upon the voltage supplied to its input, and which directly supplies the bias voltages for opening or closing the gate in the carrier voltage path).

A photograph of a single channel ring modulator of this type is shown in Fig. 4. The controls on the front panel are the threshold control for the

adjustment of the minimum program level for opening the carrier gate, the squelch on-off switch for activating or de-activating the carrier suppression circuit, the ring modulator balancing adjustments (which normally remain untouched), and the pilot light and switch of the built-in power supply unit.

Among all signal processors the multiplier-type ring modulator takes a unique position, since it is capable of converting existing sounds into new (and pleasing) sounds with entirely different overtone spectra that do not resemble the original acoustical phenomena. A few examples will illustrate some typical applications of this sound processing tool and the results obtained.

Ex. 1: A 1000 Hz sine wave is applied to the program input and a 900 Hz sine wave to the carrier input. The output contains two frequencies, 100 Hz and 1900 Hz. If the magnitudes of the inputs are both 1.0 volt RMS, the magnitude of the total output of the described standard model will also be 1.0 volt RMS.

Ex. 2: The program input receives a 1000 Hz square wave and the carrier input receives a 900 Hz sine wave. A square wave contains an infinite series of discrete frequencies, all of which are odd multiples of the fundamental. The output therefore consists of two infinite series, one of which is the sum of the 1000 Hz square wave components and the 900 Hz sine wave, and the other of which is the difference.

Ex. 3: Program input is filtered white noise with a bandwidth of 0 to 100 Hz and carrier input is a 900 Hz sine wave. This noise spectrum contains equal energy per unit bandwidth from 0 to 100 Hz. The output of the modulator is a spectrum centered at 900 Hz, but containing an equal distribution of frequencies from 800 Hz to 1000 Hz. Note that the bandwidth of the output is twice the bandwidth of the program input. When sweeping the carrier frequency of this setup over the center portion of the audio range, the sound of a howling wind may be simulated. A similar but more complex effect will be obtained when the program input is white noise with a bandwidth of, for instance, 400 to 500 Hz. In this case a carrier of 900 Hz would generate two white noise bands, one from 400 to 500 Hz and one from 1300 to 1400 Hz. Naturally, "tuned" white noise may cover a lesser bandwidth and thereby result in more selective effects.

Ex. 4: The program material is supplied by a voltage-controlled oscillator which operates in the sine wave mode and is controlled by a keyboard. The carrier signal is supplied by a second voltage-controlled oscillator in the sine wave mode, controlled by the same keyboard and tuned relative to the first oscillator by a frequency ratio of 3:4, or any rational number. In case these integers do not have a common denominator, the resulting fundamental frequency and its overtones at the output will be of a very attractive quality due to slow timbre changes, which may result from an intentional detuning of the two input frequencies relative to the theoretical multiples of the fundamental frequency.

Ex. 5: Very interesting effects with the speaking or singing voice may also be obtained by feeding the fundamental voice frequency (obtained through

a lowpass filter) into one input and the entire voice spectrum into the other. In this case the application of an efficient automatic gain control to the fundamental frequency (with the aid of a voltage-controlled amplifier) would be required, in order to retain the original dynamic properties of the input sounds.

Ex. 6: When feeding the program material (preferably music and very effectively, organ music) into the carrier input and a low frequency sine wave in the vibrato range (for instance, 6 Hz) into the program input, a special modulation effect will be created, and will be remarkably enhanced if the same program material is reproduced directly (without modulation) through a second amplifier and speaker system. The result will be a kind of spatial amplitude-phase modulation.

Ex. 7: Percussive sounds in the category of Trinidad drums are obtained when the sounds of bass drums, tom toms, temple blocks, wood blocks, claves, and maracas are fed into the program input and an audio frequency in the lower to middle audio range into the carrier input.

Ex. 8: When the program material is heterodyned into a higher frequency range, say 10,000 to 20,000 Hz (with the aid of an oscillator of appropriate frequency feeding into the carrier input), and the new spectrum is passed through a narrow band filter in said frequency range, and the filtered frequencies heterodyned back into the audio range by applying the same oscillator frequency to the carrier input of a second ring modulator, the effect of a tunable filter is obtained when the oscillator frequency is changed.

From these examples, which merely scratch the surface of the possible applications of the multiplier-type ring modulator, it will become evident that this instrument is a very powerful tool for the electronic music composer, and that the variety of results obtainable is as limitless as the imagination of the user.

Notes on Mixtur[1,2] (1964)

Karlheinz Stockhausen

In my composition *Kontakte*[2] for electronic sounds, piano, and percussion (1959-60), the electronic music I had arrived at in the studio was put on a four-track tape. It was then reproduced through loudspeakers while a pianist and a percussionist played. After composing *Kontakte*, I looked for new possibilities of directly but flexibly joining electronic sound production with instruments, and of transforming sound with electronic equipment. I experimented in this direction.

The first result was *Mikrophonie I*[2]: in a live performance of this score a gong approximately five feet in diameter is made to vibrate by two players using various materials; at the same time, two other players move microphones over the surface of the gong; the vibrations the latter pick up are altered by two more players with electro-acoustical filters and potentiometers; the result is simultaneously reproduced via two loudspeakers.

In *Mixtur* the sounds of a woodwind ensemble, a brass ensemble, and two string ensembles—one pizzicato—(seated in four groups around the audience) are picked up by microphones and put into four ring modulators; the four groups of microphones lead to four mixing tables, where sound engineers control the balance of the various microphones and the input levels for the ring modulators (during the last public performance of *Mixtur* at Stockholm in October 1966, a total of 36 microphones was used, one microphone for each stand with two musicians). Four players, each using a sine wave oscillator with continuous frequency control, produce sine waves with which the instrumental sounds are modulated by the ring modulators. The results, reproduced over four separate loudspeakers, are blended with the orchestral sound. From each instrumental sound there arises a Mixtur-sound. (By "Mixtur," one usually refers, regarding organ stops and also choral and orchestral melodies, to a mixing of parallel pitches. It is then a matter of timbral texture from overtones or parallels of chromatic intervals.) The fifth instrumental group of *Mixtur*, consisting of three percussionists each playing a cymbal and gong, is provided with contact microphones connected to three separate loudspeakers. So a composition of differentiated timbres—which I had heretofore been able to achieve only with electronically produced sounds—becomes possible with the use of instruments.

In addition to the various transformations of timbres, it is possible to compose with as subtle differences in pitch as may be desired, beyond the common division of the octave into twelve equal steps. A rhythmic transformation of

[1] First performed Nov. 9, 1965 in the series "Das neue Werk" of Norddeutscher Rundfunk, Hamburg.
[2] The scores of *Kontakte* (realization-score and performance-score), of *Mikrophonie I*, and of *Mixtur* are published by Universal Edition, Wien (available in the U.S. through Theodore Presser Co., Bryn Mawr, Pennsylvania).

the instrumental sounds occurs whenever these sounds are modulated with sine wave frequencies lower than 16 Hz.

It seems to me that the future development of instrumental music is completely open; the indispensable characteristics of instrumental music (above all, its changeability through history, its very liveliness) are brought to a new unity with the achievements of electronic music, a unity which is incomparably more flexible and more capable of change than the combination of tape and instrumental music of recent years.

There is no doubt that such a procedure does demand a completely new style of composing, of shaping, of notation. The challenge presented by *Mixtur* was to me a welcome inducement, an invitation, to keep my mind perpetually open for unimagined, unheard things. I wrote the score rather quickly, and without interruption, during the summer of 1964; only inspiration was obeyed, as experience was lacking. It is my hope, however, that the music retains something of the freshness and happy mood of those adventurous days.

[Translated by William Sylvester]

Beginning January 1967

SOURCE

music of the avant garde

Unique periodical, presenting in each issue five provocative new works attractively published in open score plus related articles, photo essays, and interviews.

Representative contributors include Robert Ashley, Larry Austin, Earle Brown, Barney Childs, Pauline Oliveros, Harry Partch, David Reck, William O. Smith, Karlheinz Stockhausen, Bertram Turetzky, David Tudor.

One year, 2 issues, $9.
Three years, 6 issues, $25.
Single issue rate, $5.

Write to:

SOURCE
330 University Ave.
Davis, Calif., 95616

Electronic Music Three Ways

Robert Ceely

Though vigorous pioneering efforts in electronic music did exist in the United States throughout the nineteen-fifties, it is certainly true that the real impetus for this new music was European. It is also true that since about 1960 America has been in a very real sense turned on to electronic music. This is evidenced not only by the proliferation of studios—large and small, university and independent—but also by the general public's awareness (if not always acceptance) of this music through the rash of concerts, radio broadcasts, and popular articles. Nor has the recognition of America's awakening to electronic music been merely domestic. The lectures by Lejaren Hiller at Darmstadt in 1963; those by Milton Babbitt in 1964; the appearance of the Ann Arbor "ONCE" group at the Venice contemporary music festival in September 1964; the growth of European radio and concert performances of much American electronic music: all have served notice to Europeans (more especially the European composers) that, while they may not like the electronic music being composed in America, they can no longer pretend it does not exist.

European electronic music studios were created and exist now as part of government-owned radio stations. (As rare exceptions, one thinks of the studio at the Siemens company in Munich, the university studio at Utrecht, the independent studio of the late Hermann Heiss at Darmstadt.) The advantages for a studio contained within a large radio station are many. Much of the equipment necessary to initiate a studio is already at hand, temporary replacements of faulty equipment are readily available, and technicians are in abundance to repair and maintain the studio equipment. Government ownership of the studio means that its use and all necessary materials (from recording tape to splicing tape) are free to the composer. Other advantages are more personal, perhaps idiosyncratic, and relate only to my own experience in the Milan studio: the well-stocked bar on the floor above, the lovely girls one meets on the elevator, and the uniquely gifted resident technician of the studio, Marino Zuccheri.

Many in Europe as well as America believe that the electronic studio situation here compares favorably to the situation abroad: although the European radio studio offers all materials free, the American university essentially goes much further by stocking its studio with much more sophisticated equipment and offering composers fat fellowships and/or professorships, enabling them not only to buy tape but groceries as well. If one wishes to completely ignore the very significant phenomenon of the independent "home" studios and to offer as examples only the Columbia-Princeton studio and the University of Illinois studio, then the American studios are indeed an improvement over the European in many respects. If one reads "synthesizer" for "sophisticated equipment," one American studio (Columbia-Princeton) is indeed "better" than all those European studios put together. And since most are resident

or visiting faculty, the composers are, in a manner of speaking, completely supported while working in these two studios. Unfortunately, all the young, non-university composers who have in any real sense been directly subsidized to work in either of these studios have been from outside the United States.

Most of the other, emerging university studios are either too new or too small for one to adequately discuss their present status or to risk predicting their future importance. However, one is saddened to notice certain oddities surrounding the births of several of these new studios. One should hope these aspects will not be imitated by other beginning studios and thus become syndromes of all university studios. One can already observe: (1) the 'Ampex syndrome:' a studio must have two or three Ampex 350 tape recorders immediately no matter how small its initial budget or how much it needs other, more basic equipment; (2) the '"they've got one we've gotta have one" syndrome:' the desire for a studio does not satisfy the artistic need of any faculty member but merely is wanted as necessary armament in an inter-university power struggle; and (3) the 'locked studio syndrome:' the studio has been "built" by an outsider, a visiting famous composer experienced in the composition of electronic music but untainted by any knowledge of electronics. The resident faculty fools around for a while and then appoints the youngest, oldest, or newest faculty member as the director. He locks the studio, declaring it ("temporarily") off limits to all students, with the official statement that the *Music* Department is preparing a special sequence of courses enabling students to work in the studio, but with the private fears that a student composer will out-compose him electronically or—worse still—wander into the studio and ask the director to define "impedance."

To suggest that other new university studios will, through ignorance or arrogance, merely repeat and compound—and thus codify—the mistakes of some is to be pessimistic rather than realistic. By chance or design it is certain that important, vital studios will flourish within some universities. At the same time one suspects: that the true value of the university studio will ultimately prove to be pedagogic rather than artistic; that many university studios, once established, will be short-lived as interest shifts toward computers as a solution to compositional (as well as musicological) problems; and, that even in universities where the studios remain continually active, there will be the enigma of where the student composer is to compose after he leaves the university. It is not uncommon now for a graduating composer to be unable to secure a teaching position; if an electronic composer, he will more than likely find himself not only without a job but without a studio as well. The composer must then ponder three choices: postpone his desire to compose electronic music; apply for a fellowship to work in a European studio; or, investigate the possibility of building his own independent studio.

The circumstances of the independent electronic composer are in many ways analogous to those of the independent film maker: both need relatively expensive equipment for their work; both usually have limited budgets and thus cannot afford to commit errors in purchasing equipment; and, both (though often finding a sympathetic audience among university students) are

usually not themselves members of the academic community. (That the independent composer often supplies electronic soundtracks for the film maker, and joins him in cooperative festival-type events, is interesting and perhaps prophetic.) It is paradoxical that, while one can easily spend a great deal of money building an electronic studio and not be extravagant in any way, it is becoming equally true that one can spend comparatively little money and still have a remarkably versatile and productive studio. The latter may lack the sophistication and some of the potential of the university studio, but is often superior in ease of operation and, of course, in availability. Since the equipment in the independent studio will be used only by one or two composers, it need not be as durable or as expensive as that used in a university studio. Recent progress in the development of devices designed especially for the composition of electronic music means that the independent composer can obtain relatively inexpensive equipment fitting his individual compositional needs. Information regarding the construction of independent studios is becoming increasingly available, both in published articles[1] and in private communications.

One suspects that the growth of independent studios is not due solely to the dearth of university studios with "space" for non-aligned composers, nor is the independent studio merely a stop-gap contraption which will wither away in that Utopian future when every electronic composer has the use of an institutional studio. Rather, one foresees the continual usefulness of the independent studio. To the composer temporarily (or permanently) "without studio" it will always be a necessity, to the instutional composer it will provide a domestic device for at-home composing, and for the enlightened layman its advantages over a Hammond organ should prove to be immeasurable.[2]

Despite, indeed because of, the increase in both independent and university studios, there is already the need for a third type of studio. While having advantages of both the independent studio (it would not be connected with a university) and the institutional (it would be similarly equipped), it would most clearly resemble the best aspects of the "classical" European studio. Electronic music is too important to leave to the university, and to entrust its future solely to the independent composer is to demand too much of him

---

[1] Gordon Mumma, "An Electronic Music Studio for the Independent Composer," *Journal of the Audio Engineering Society,* XII, 3, July 1964, 240.

[2] Arnold Schoenberg in a letter to Dr. Werner David, May 10, 1949:

> Therefore, I believe that the instrument of the future will be constructed as follows: there will not be 60 or 70 different colours, but only a very small number (perhaps 2 to 6 would certainly be enough for me) which would have to include the entire range (7-8 octaves) and a range of expression from the softest pianissimo to the greatest fortissimo, each for itself alone.
>
> The instrument of the future must not be more than, say, 1½ times as large as a portable typewriter. For one should not strike too many wrong keys on a typewriter either. Why should it not be possible for a musician, also, to type so accurately that no mistakes occur?
>
> I can imagine that, with such a portable instrument, musicians and music-lovers will get together in an evening in someone's home and play duos, trios, and quartets; they will really be in a position to reproduce the idea-content of all symphonies. This is, naturally, a fantasy of the future, but who knows if we are so far away from it now?

The Works of Arnold Schoenberg, ed. Josef Rufer, trans. Dika Newlin, Faber and Faber, London, 1962, 68.

and his equipment.

Needed to complement the two existing types of studios are government or foundation-supported electronic music studios, preferably located in large cities. Each of these studios would contain complete electronic sound-production facilities as well as additional rooms for mixing and splicing. Each studio could thereby accommodate several composers engaged in various compositional activities. Each studio would be directed by an American electronic composer and the salaried staff would consist of a technician and a secretary. The use of studio facilities and all materials would be free to participating composers. Interested composers would apply to the director of each studio for admission and, if in need, would be eligible for financial aid to sustain him while working in the studio. The length of time spent at a studio would vary with the type of project the composer would wish to undertake, and be limited only by the number of applicants and studio availability. While the director would arrange concerts of work done in the studio, and in general advertise its activities, the studio would not aim to be educational. Nor would it ever hope or plan to be self-sustaining; its existence would at all times be dependent upon government or foundation support.

RECORDED SOUND

Quarterly Journal of the British Institute of Recorded Sound

Recorded Sound regularly contains book reviews and discographies and also articles on the music of the twentieth century, for example:

Nos. 5, 6, 15	JANE BATHORI: Reminiscences of Debussy, Ravel, Fauré, Roussel, and later French composers
No. 8	VARIOUS AUTHORS: Items on Busoni; GIOVANNI MARTINELLI: My association with composers of the verismo school
Nos. 13, 14	ERNEST ANSERMET: The crisis of contemporary music
No. 14	HUGH DAVIES: Discography of electronic music and musique concrete
Nos. 22/23*	Supplement to the same
No. 17	MICHAEL TIPPETT: Music and poetry
No. 18	PIERRE BERNAC: Poulenc's songs
No. 20	ANDRZEJ PANUNIK: Karol Szymanowski
No. 21	ANGUS MORRISON: Recollections of Hahn
No. 24	MARCELLA BARZETTI: A meeting with Nono
Nos. 24, 25	RICHARD STRAUSS: Notes on the interpretation of Beethoven's symphonies

Single number $1; *double number (22/23) $2; post free.
Subscription (4 numbers) payable 1 January $3.50
Write to: British Institute of Recorded Sound,
29 Exhibition Road, London, SW7, Great Britain

Symposium: Programmed Control

Introduction to Programmed Control

Robert A. Moog

Most electronic music has been produced largely by "classical" studio technique, by which individual sounds are recorded on magnetic tape and then assembled into the finished composition by tape editing and mixing. With more sophisticated equipment now available, it is appropriate to consider techniques by which the parameters, including the timing and duration, of an entire collection of sounds are specified by the composer prior to the recording operation. These techniques are known collectively as programmed control, and are capable of saving the composer as much as 90% of his time in realizing a piece of electronic music, while simultaneously enabling him to assemble precise and intricate sound structures.

The basic concepts of programmed control in music are certainly not new. Pneumatically activated polyphonic tone generators programmed by multichannel perforated paper tape (otherwise known as player pianos) predate the phonograph. Other programmed "music machines," based on the clock mechanism, date back to the 13th century. Perhaps the best known instruments designed for programmed electronic music composition are the R.C.A. Synthesizers.[1,2] These large instruments incorporate a complete array of analog signal generating and processing circuits, and are programmed in 4-bit binary codes on special 40-channel perforated paper tape. The original claim that ". . . the electronic system can reproduce or create any sound or combination of sounds, which have or have not been produced, that may have any possible musical significance."[3] now seems to be subject to qualification. However, the programming techniques used by R.C.A. are basic in nature, and have been adapted to much recent work in this field. The contributions to this symposium describe current trends in programming techniques and instrumentation. All of the articles are introductory in nature, and we can expect much more work to be done before the "ultimate" programming schemes have been achieved.

Four general classes of apparatus for programmed control are now available and have been successfully utilized. In order of increasing elegance, programming capacity, and cost, they are: the sequencer, the punched paper

---

[1] Harry F. Olson and Herbert Belar, "Electronic Music Synthesizer," *Journal of the Acoustical Society of America*, XXVII, 3, May 1955, 595.

[2] H. F. Olson, H. Belar, and J. Timmens, "Electronic Music Synthesis," *Journal of the Acoustical Society of America*, XXXII, 3, Mar. 1960, 311.

[3] Olson and Belar, p. 595.

tape reader, the hybrid system consisting of a small digital computer which controls analog sound generating and modifying apparatus, and finally, the large digital computer capable of generating, with the aid of a digital-analog converter, an entire musical composition from a set of coded instructions.

A sequencer is a device that initiates events, one at a time, in a predetermined order or sequence. The sequence may be generated by selective gating of continuous sounds from many sources, or by changing the parameters of the sound from a single source, or by performing both functions. Generally, sequencers are capable of automatic progression from event to event, and have provision for setting the durations of the events. Thus a sequencer is capable of eliminating as many tape splices as the number of events (typically eight) it can initiate without repeating. The composer has direct and immediate control over all sound parameters, and can repeat a sequence after adjusting the controls until he obtains the desired result. Sequencer programming is generally accomplished "by ear" and does not lend itself to the precise and permanent programming that is characteristic of the other three methods.

A punched paper tape reader detects either the presence or the absence of holes along lines on the tape being fed through the reader. In its simplest mode of operation, the presence of holes along a line will trigger events, just as a player piano roll does. More sophisticated programming schemes involve the use of digital codes, and the simultaneous reading of several parallel lines on the tape. With these more complex schemes, a tape reader may be capable of rapidly and accurately controlling several parameters of sounds produced by conventional voltage-controlled generating and processing equipment. Although the idea of paper tape readers seems to connote highly mechanized data processing, paper tape is an appropriate programming aid to the creative process. The controlled analog generating and processing equipment may be manipulated by the composer while the tape is being run. The tape itself may be manipulated with the same, if not more, ease than magnetic tape, and can be decoded on sight by the composer almost as easily as a conventional score. The main disadvantage of punched paper tape programming is the relatively long time required to prepare the tape. Another drawback might be the speed that the tape is capable of moving, but this depends mainly upon the type of tape reader being used.

The hybrid digital computer / analog sound generating system overcomes the disadvantages of the paper tape reader. Here a small digital computer replaces the paper tape reader as the device which issues commands to the analog generating and processing equipment. The composer communicates his instructions to the computer in a simple, easily-understood notation, and the computer converts the instructions into the appropriate electrical signals and conveys them to the sound generating and processing instruments. A disadvantage of this method, as compared with paper tape programming, is that a computer is harder to physically manipulate than a segment of paper tape and the composer gives up another measure of direct control in exchange for increased programming efficiency.

In the three programming methods described so far, the programmed devices are generally analog voltage-controlled signal generating and processing instruments such as are found in many contemporary studios.[4,5] The last programming method on our list dispenses with analog signal generating and processing instruments entirely, and computes the sound material itself (not merely commands to auxiliary sound-generating instruments). A large digital computer and a digital-analog converter are required for this type of programming.[6] The spectrum of sounds which can be programmed is now no longer limited by the auxiliary signal generating and processing instruments, but is determined solely by the nature of the program. Thus the information that the program must specify is much greater, and no intervention by the composer between the writing of the program and the completion of the composition is possible.[7]

It should now be obvious that each of the four programming techniques has both advantages and disadvantages. The "best" system depends very much upon what type of composition is contemplated, as well as upon the inclination of the composer. While many composers are at one with their splicing blocks, others seem to be equally at home operating a keypunch, producing a punched paper tape, or manipulating the multitude of control knobs on a sequencer. The following three papers present various aspects of paper tape and computer programming in some detail. Emmanuel Ghent describes several ways of using paper tape, especially to control voltage-controlled instruments. He also discusses the problem of tape preparation. George Logemann discusses the use of computers in preparing paper tape and as programming devices. James Gabura and Gustav Ciamaga describe a simple and easily understandable system consisting of a small digital computer and a modest array of voltage-controlled instruments.

---

[4] Robert A. Moog, "Voltage-Controlled Electronic Music Modules," *Journal of the Audio Engineering Society,* XII, 3, Oct. 1965, 200.

[5] R. A. Moog, "A Voltage-Controlled Lowpass Highpass Filter for Audio Signal Processing," Audio Engineering Society Preprint no. 413, from 17th Annual Meeting, Oct. 11-15, 1965.

[6] James Tenney, "Sound Generation by Means of a Digital Computer," *Journal of Music Theory,* VII, 1, Spring 1963, 24.

[7] With modern graphic input-output devices such as the light-pen/cathode-ray tube system, the composer is able to change his program by redrawing in a manner similar to that which he would employ to change a conventional score.

Comparison of

	Sequencer	Paper Tape
ADVANTAGES	1. Offers the composer immediate control over all sound parameters, and may be easily changed during programming. 2. Unlimited speed of operation.	1. Combines potentially high accuracy with ease of manipulation. 2. Unlimited program length. 3. Tape may be read like a score. 4. Control of polyphonic material is feasible. 5. Equipment is relatively simple and inexpensive.
DISADVANTAGES	1. Length of program is limited by size of sequencer. 2. Limited accuracy of programming.	1. Reading speed of mechanism may be limited. 2. Preparation of paper tape tends to be time-consuming.
COST RANGE	\$20-\$50 per programmed event, plus appropriate analog equipment.	\$500-\$5000 complete, plus appropriate analog equipment.

Programming Equipment

Computer Control of Analog Equipment	Computer Generation of Audio Material
1. Program preparation is highly efficient. 2. Analog equipment may be manipulated while program is being run. 3. Accuracy of programming is limited only by auxiliary analog equipment. 4. Program can include computation of simple control functions.	1. Audio signal parameters are not limited by auxiliary analog equipment. 2. Use of graphical input-output devices allows the composer to literally draw scores for realization by the computer.
1. Difficult to manipulate program itself while being run. 2. Equipment cost is high.	1. Composer has no control of audio parameters while the program is being run. 2. Cost of using equipment precludes the feasibility of casual experimentation.
\$25,000-\$50,000 complete, plus appropriate analog equipment. Small digital computers can also be rented.	Equipment rental costs are typically \$100 per minute of audio material.

Definitions

An electrical voltage may be constant in magnitude (called DIRECT or DC) or varying about zero magnitude (called ALTERNATING or AC).[8] The POLARITY of a direct voltage is the direction of the electrical force that the voltage produces, and is always referred to a convenient REFERENCE or BASELINE voltage. The common baseline voltage for a system is called GROUND voltage, but other baseline voltages usually exist locally within a system. A QUANTUM of voltage is a small voltage difference, the magnitude of which is fixed.

A FUNCTION is the pattern of the variation of a quantity with respect to another changing quantity, usually time. A LINEAR function is a function in which the dependent variable (voltage in this case) changes in direct proportion to any change in the independent variable. It is the only function that is graphically represented by a straight line. An EXPONENTIAL function is a function in which the dependent variable changes by a fixed ratio whenever the independent variable changes by a fixed absolute amount. A LOGARITHMIC function is the inverse of an exponential function.

A SINE WAVE is a PERIODIC or regularly repeating function with only one frequency component. A FILTER is a signal processing device which reinforces certain frequency bands while attenuating others.[8] A BANDPASS FILTER reinforces one frequency band and strongly attenuates all other frequencies. The ENVELOPE of a signal is the contour of the amplitude of the signal versus time. An ENVELOPE GENERATOR generates a slowly-varying non-repetitive function (voltage) which is subsequently used in a system of voltage-controlled instruments to impart an envelope to a steady signal. A GATE is a switching circuit that passes or shuts off a signal in response to a control or GATING VOLTAGE. A LIMITING AMPLIFIER (sometimes called an amplitude filter) modifies the envelope of the signal, usually by limiting the maximum amplitude or by sharply attenuating the signal whenever the amplitude falls below a certain preset level.

A POTENTIOMETER or "POT" is an electromechanical device, the output of which is a certain percentage of the input. The WIPER ARM of a potentiometer is its moving element, the position of which determines the ratio of output to input voltage.

A RELAY is an electromechanical device that switches one or more voltages upon the application of an energizing voltage. It is the electromechanical counterpart of the gating circuit. A LATCHING RELAY is a relay which remains in the energized state even after the energizing voltage is removed, and must be de-energized (unlatched) by the application of a second de-energizing voltage. Latching relays, or their wholly electronic counterparts, can be the basic components of a digital MEMORY since they retain their states indefinitely. A MERCURY-WETTED RELAY is a relay whose switch contacts are wetted with mercury, and are therefore able to switch extremely rapidly with

---

[8] For more detailed definitions of DC and AC, and for terms involving frequency and frequency bands, see the introduction to Harald Bode's paper in this issue of EMR.

a minimum of extraneous switching noise.

A COMPUTER is a device which receives information, usually in coded form, then processes it in a prescribed way and delivers it for further use. A DIGITAL COMPUTER processes information by counting electrical pulses; an ANALOG COMPUTER processes information by measuring the magnitudes of electrical voltages. A HYBRID COMPUTER utilizes both digital and analog computation. A device which converts information in digital form to essentially the same information in analog form is a DIGITAL-ANALOG CONVERTER or DIGITAL-ANALOG INTERFACE. The inverse function is performed by a device called an ANALOG-DIGITAL CONVERTER or ANALOG-DIGITAL INTERFACE. A simple, commonplace analog-digital converter is a gasoline pump, which measures the amount of gasoline and then states the number of gallons in discrete digital units.

An analog computer consists of an array of signal generating and processing instruments. Programming is accomplished by specifying both the modes of operation of the instruments and the interconnections (patches) between them. An instrument input to which programming information is fed is called a CONTROL INPUT, while the voltage bearing the programming information is called a CONTROL VOLTAGE. A FUNCTION GENERATOR is any generating instrument—it may or may not be programmable. Both oscillators and envelope generators are function generators.

A SYNTHESIZER, as used here, is a specialized analog computer-like instrument for generating audio material; it offers independent and convenient control over the important sound PARAMETERS, the variable quantities that are measurable.

An INPUT of a device accepts information either for processing by that device, or for determining the mode of operation of the device. Information for processing by a computer is known as DATA, while instructions specifying how data is to be processed are known as a PROGRAM. Information which controls the modes of operation of analog instruments is also known as a program. A SUBROUTINE is a logical subsection of a program embodying a distinct and complete sequence of operations. A LOCALLY DEFINED SUBROUTINE is a subroutine which is not a permanent part of a program, but is written by each user for his own application. The processed data (in the case of a digital computer) or the processed signal (in the case of an ensemble of analog instruments) appears at the OUTPUT of the device.

INPUT-OUTPUT DEVICES of a computer convert written, recorded, or otherwise coded information to appropriate sequences of electrical pulses, or vice versa. Typewriters, paper tape readers, digital magnetic tape recorders, and keypunch machines are all input-output devices. An input-output device that is connected directly to the central processing unit of a computer is said to be ON LINE. A DRUM CARD is a simple program for a keypunch machine, and is directly analogous to the tab stops on a typewriter. Information entered on punched cards without regard to the columns on the cards is FREE FIELD. An ADDRESS is a location in a computer or other signal-processing device to which information is to be routed. MULTIPLEXING is an arrangement by which

REPERTOIRE INTERNATIONAL DES MUSIQUES ELECTROACOUSTIQUES / INTERNATIONAL ELECTRONIC MUSIC CATALOG

Under this title the Groupe de Recherches Musicales of the French Radio is preparing a new version of the *Repertoire International des Musiques Experimentales* (1962). This new version, which is being coordinated by Mr. Hugh Davies, is taking a slightly different form from that of the original *Repertoire,* in that it is intended as a complete survey of all electronic music ever produced.

It is hoped to achieve a high level of completeness: with an expected total of over 4000 entries, the present rate of expansion of this medium will preclude the possibility of such a project ever again being undertaken. The survey, which covers the first twenty years of the existence of electronic music, is intended as a source of information for everyone connected with new music: composers, performers, conductors, teachers, programmers of public and broadcast concerts, electronic music studios, engineers, libraries, students, and critics.

In addition to including compositions produced in private studios (even those with an improvised assortment of equipment), which account for a considerable portion of the entries, certain other features not in the original *Repertoire* are included: the discography is much more exhaustive, and information on published scores and tapes which can be hired from music publishers is added; other appendices include information on predecessors of tape music (use of disc recordings, music drawn on film sound-

tracks), use of computers and synthesizers in connection with the composition and production of electronic music, tape compositions by poets, and electronic sound equipment used by sculptors; also added is a directory of studios that includes address, personnel, policies, special equipment, acceptance requirements and working conditions for composers, opportunities for study, and future plans; the catalog concludes with an index of composers, with nationality and year of birth.

The main body of the documentation defines the status of the studios and the function of each work as precisely as possible; it also shows the very early development of tape music throughout the world in much clearer detail than has ever been done before, and includes much information previously unknown. "Electronic instrumental" music (using live electronic transformation of performed music) is also included.

This catalog is a cooperative publication of Le Groupe de Recherches Musicales de l'O.R.T.F. and the Independent Electronic Music Center; it will appear as the April issue of EMR.

EMR subscribers will automatically receive a copy; single copies $2.50 (foreign $2.75—payable in U.S. dollars); order from:

Any electronic music composer or studio that has not been contacted in connection with this catalog is asked to write Mr. Hugh Davies immediately, at the following address:

ELECTRONIC MUSIC REVIEW
Trumansburg, N. Y. 14886, U.S.A.

two or more channels of information are processed or transmitted simultaneously, then separated and routed to their respective addresses.

Virtually all digital devices use BINARY counting, in which a digit may have one of only two values or states. These are generally written as 0 and 1. A digit of a binary number is called a BIT, and a binary number consisting of a specified number of digits is called a CHARACTER. A BLANK CHARACTER is a character, all of whose digits are 0.

A CODE or LANGUAGE is a set of relations for representing information with a limited set of SYMBOLS, the letters, numerals, etc., that may be typed or keypunched. Codes which use standard typewriter symbols are called ALPHANUMERIC, and those which people easily understand by association are called MNEMONIC.

PUNCHED (or PERFORATED) PAPER TAPE is a medium of digital information storage and retrieval. Characters are punched laterally and are commonly eight bits long, but may be more or less depending upon the number of tracks on the tape and the manner in which the holes are detected. The simplest form of tape reader mechanism has one line of electrical switches, each of which is under one row of holes and closes when a hole passes over it. The bank of switches is called the SWITCHING MECHANISM. More sophisticated tape readers, called BLOCK READERS, have as many as 24 lines of switches and are capable of reading all of the information in a rectangular area of tape simultaneously. A tape reader includes, in addition to the switching mechanism, a TRANSPORT MECHANISM which moves the tape past the switching mechanism at a predetermined speed. A transport mechanism may be driven by a variable-speed motor, the speed of which is controlled by a MOTOR-CONTROLLER system. A TACHOMETER, or rotary speed meter, attached to the tape drive sprocket, indicates how fast tape is moving, and a COUNTER, attached to the same drive sprocket, indicates what portion of the tape has already passed the switching mechanism.

A CODING SCHEME is the manner in which punches from a paper control tape are interpreted and gated to the synthesizing apparatus; a SYMBOLIC CODING SCHEME is a system of symbolic notation for representing characters.

Symposium: Programmed Control

The Coordinome in Relation to Electronic Music

Emmanuel Ghent[1]

The coordinome was so christened because of its original function—to precisely coordinate performers playing at independent tempi, meters, beat positioning, and spatial location, and to accurately synchronize live performance with tape music. As the compositional significance of this application

[1] I would like to express my deep gratitude to Dr. Robert A. Moog of the R. A. Moog Co. and Mr. Stein G. Raustein of New York University for their invaluable technical advice and assistance. Mr. Raustein helped enormously in the design of the coordinome itself and Dr. Moog must take full credit for its application to programming voltage-controlled equipment.

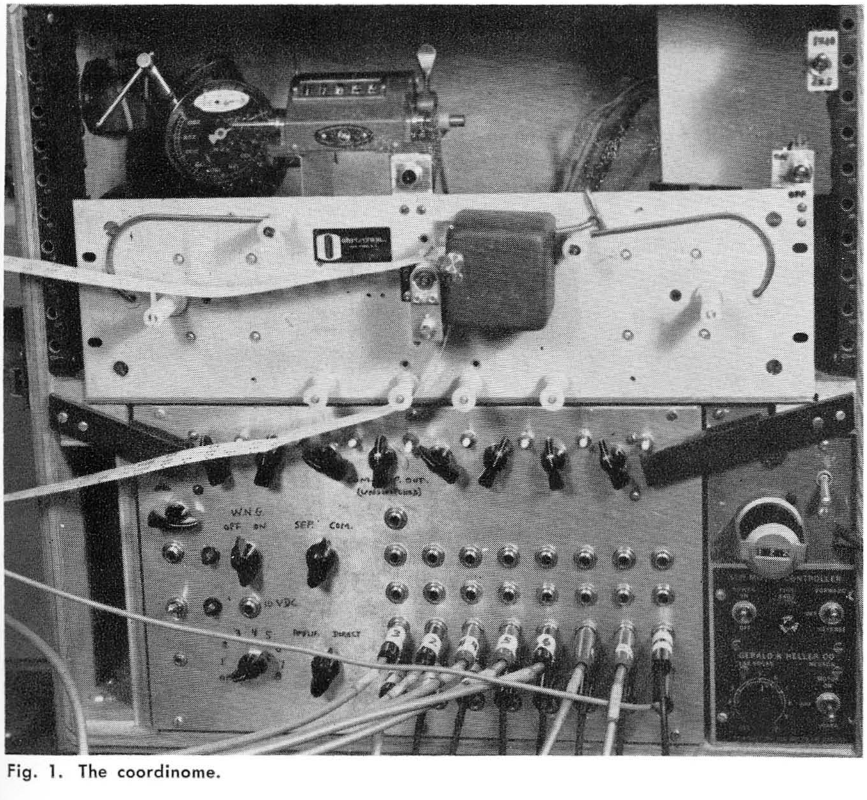


Fig. 1. The coordinome.

has been discussed elsewhere,[2] it is the principal purpose of the present paper to outline a new adaptation of the device to the needs of electronic music.

Stripped of its auxiliary equipment, the coordinome (Fig. 1) is in essence a punched tape reader driven by a variable speed motor. The switching mechanism of the reader (eight pole, single throw, common ground) is triggered by the holes in the perforated tape. As there are eight bits per character or vertical line of tape, the reader activates eight mercury-wetted relays (or transistor switches), which in turn gate up to eight independent input signals. Depending on the particular application, the input will be either audio signals from oscillators, tape recorder outputs, etc., or DC voltages. Audio signals may be switched directly or passed through the eight built-in amplifiers before being switched (Fig. 2). If a common input is being used, individual amplification of each channel before switching is usually necessary in order to counter the variations in load that would be occasioned by simultaneous gating of several channels. DC voltages are always switched directly.

The tape transport is sprocket-driven and is powered by a reversible DC motor with a speed-controlling system, permitting one to continuously vary the speed over the range of 2.5 to 20 i.p.s. (25 to 200 characters per second). Both a tachometer and a suitably calibrated footage counter are included for control purposes.

[2] Emmanuel Ghent, "Programmed Signals to Performers: A New Compositional Resource," to appear in *Perspectives of New Music*, 1967.

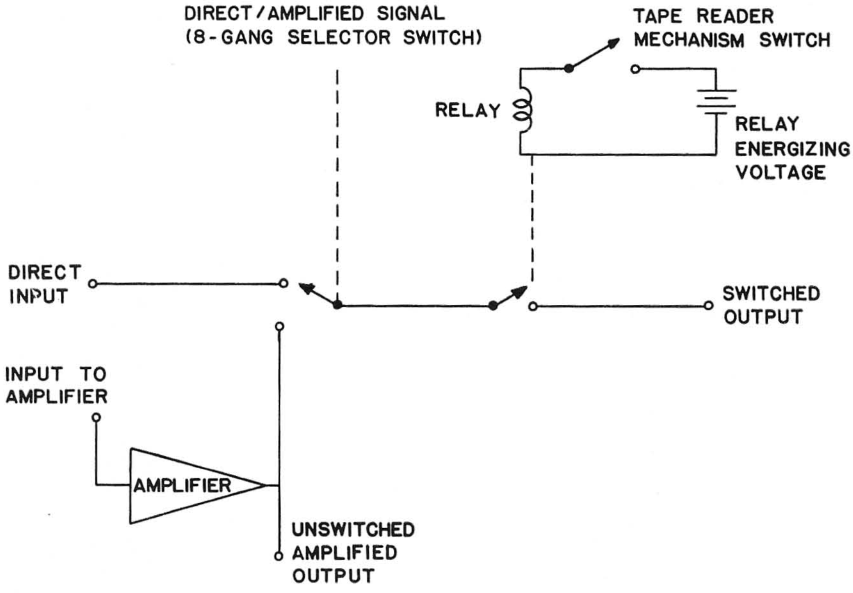


Fig. 2. Block diagram of coordinome (one of eight sections).

The perforated tape may be prepared by a variety of means, from hand-punch to computer.[3] A simple and convenient approach is to appropriately mark or punch data processing cards such as Port-a-Punch or mark-sensing cards. These then are automatically converted by machine to punched tape.

Although the focus of this paper is the use of the coordinome in controlling sound generating equipment, it will be helpful by way of orientation to outline its principal applications.

Distribution of signals to performers or conductor(s)
The purposes here are (1) the coordination of performers by means of periodic or aperiodic beat signals where each performer may be playing parts written in different and changing tempi, meters, etc., (2) the synchronization of performers with tape music, or (3) both types of coordination. The punched tape is prepared so that each of the eight horizontal tracks represents a separate input-output channel. A hole in channel 1 will then gate the input to channel 1 for a duration varying inversely with the speed of the tape. The current procedure is to use sine wave inputs with half-octave increment per channel: 250, 350, 500, 700 Hz, etc., respectively. The eight or fewer outputs are mixed and recorded on a single track of magnetic tape. Once the multiplexed tape has been prepared the coordinome has served its purpose. All that is required for rehearsal of performance is an ordinary tape player, the amplified output of which is distributed by cable (or ultimately, radio transmission) to the performers (Fig. 3). Decoding is accomplished peripherally by means of a filter and limiting amplifier circuit, located in the decoder box just proximate to the miniature earphone used by the performer. In a recent work[4] by the author these signals were recorded on one of four tracks of half-inch magnetic tape, while the electronic part of the composition was recorded in precise synchronization on the other three tracks.

[3] Applications of the computer to this system as well as to other aspects of electronic music synthesis are discussed by George Logemann in another contribution to this symposium.
[4] *Hex, an Ellipsis for Trumpet, Instruments, and Tape.*

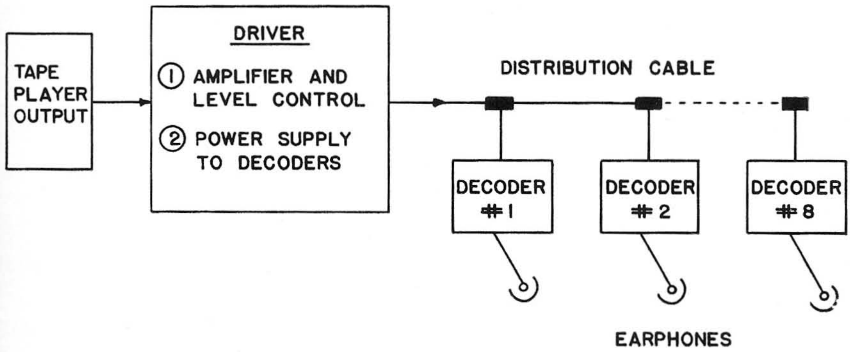


Fig. 3. Block diagram of the system for signal transmission to performers.

Use of the coordinome as a gating device in the composition of electronic music

By varying the audio signal inputs the composer may use the rapid gating capacity of the coordinome for purposes of creating electronic music fragments, which then are subjected to the usual studio procedures. Rhythm can be specified precisely, and a degree of envelope control becomes available by using the gated output signal to trigger an envelope shaper. Through the use of gates of somewhat longer duration, seconds rather than milliseconds, the coordinome may also be used during the process of composition as a means of accurately distributing electronic music materials to any pattern of up to eight speakers or eight magnetic tape channels.

Use of the coordinome for teaching purposes

Rhythmic usage in recent music has tended to make obsolete the forms of rhythmic training based essentially on binary subdivisions. Extensive use of complex additive structures with internal subdivisions, and of frequently changing proportionately related tempi (often notated as irrational subdivisions, especially when of very brief duration), are among the problems that would yield to more appropriate training. To this end a few specially programmed paper tape loops "played" on the coordinome over a wide speed range would be a valuable instructional and practice aid.

Use of the coordinome as a means of gating auxiliary functions

Where theater or dance plays a role in the production, gated signals could be used as cues to actors, dancers, or lighting engineers, or they could be made to automatically control a complex battery of theatrical lights.

Synthesis of electronic music by programmed control of sound generating and processing equipment

Over the last few years a variety of sound generating equipment has become available whose design incorporates an exponential dependence of frequency or amplitude (gain) upon input control voltage. Since pitch and loudness are exponential functions of frequency and amplitude, respectively, the result is a linear relationship between voltage and pitch, in the case of a voltage-controlled oscillator, or voltage and loudness, in the case of a voltage-controlled amplifier. The consequences of this type of relationship are far-reaching, especially as they apply equally to other aspects of sound production, such as filtering, frequency modulation, etc.

The linearity of the function makes possible the simple application of binary coding. Using pitch as an example, four bits would be used to define 2^4 or 16 tones (pitch classes), and another two or three bits could be applied to octave registration. For a twelve-step division of the octave it is usually easier to use only twelve of the sixteen possible codes for pitch class. A two-octave chromatic scale would then be encoded as in Fig. 4.

Since the applied voltage is the sum of a baseline voltage and an integral multiple of some incremental quantum of voltage q, it follows that manual adjustments of either of these control voltages will affect pitch in a significant way. A change in the baseline voltage will result in a literal transposition of the pitches as programmed. Alteration of the magnitude of the incremental

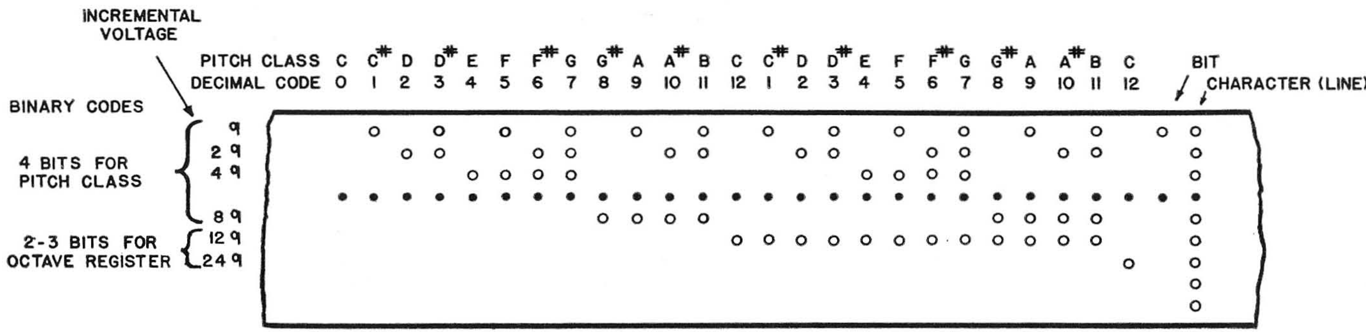


Fig. 4. Encoding of a two-octave chromatic scale.

quantum will affect the size of the interval between pitches. For instance, if the incremental quantum is halved, what was programmed as a tempered two-octave chromatic scale will now be heard as a 24-tone one-octave scale with equal quarter-tone intervals. This type of compression or expansion of any programmed intervallic sequence is possible over a wide range. Another simple operation, reversal of the polarity of the control voltage, will result in a literal inversion of whatever has been programmed. Reversal of the tape direction either by motor reversal, or by interchanging the supply and pick-up reels, will yield a literal rhythmic retrograde.

If the control voltages are fed into an oscillator controller which in turn controls any number of oscillators tuned to whatever constellation of intervals, the result will be a sequence of pitch combinations or "chords" all maintaining the identical intervallic structure. This can be as simple as, say, a line of parallel fifths, or it can be subtly used for timbral purposes, especially considering that the waveform of each oscillator may be sawtooth, variable-width pulse, triangular, or sine. The mixed output may of course then be subjected to further timbral modifications. With a sufficient number of oscillators and envelope shapers it would be easy, for example, to tune the oscillators as selected harmonics of a given fundamental, and introduce a few "non-harmonic partials" for additional color. By giving each oscillator a different envelope prior to mixing, a very rich palette of sound becomes available. Even without this relatively elaborate process a great variety of timbral coloration can be created by subjecting even fairly simple commonly-controlled oscillator output signals to envelope control, to filtering procedures in which both the center frequency and the bandwidth of a voltage-controlled bandpass filter are easily adjustable, and to frequency modulation, amplitude modulation, gain control, etc.

There is no limit to the overall length of such a sequence of pitches. The pitch range is about 10 octaves. Rhythm is programmed on an analog basis, where space on the tape is proportional to time. Absolute time is then a function of the tape speed.

A section of tape punched for the primary purpose of programming a pitch sequence as outlined above may also be used, for example, to control a voltage-controlled bandpass filter through which is being passed white noise or a low-pitched tone rich in harmonics. The resulting line of pitched noise will of course be heard as a percussive relative of that produced by the oscillator, but yet one with a specific identity.

With a more sophisticated system the door opens to vastly expanded programming. The limit on the type of programming outlined above is set by the nature of a character reader: eight bits available per event. By building memory into the conversion equipment, or by employing a block reader, a large body of information may be programmed, the useful limit now being set by the speed of the tape reader. A partial list of the variables that can be so programmed would include the following:

frequency
basic waveform[5] (sine, triangular, variable-width pulse, sawtooth)
bandpass filtering
   a. center frequency of the passband
   b. frequency width of the passband
envelope generation
   a. attack time
   b. initial descent time
   c. level of plateau
   d. final decay time
amplitude
frequency modulation[6]
   a. period of FM
   b. sweep width of FM
amplitude modulation[7]
   a. period of AM
   b. sweep width of AM
ring modulation[8]
output 1-4 channels[5]

It is also entirely feasible to simultaneously synthesize two or more completely independent and overlapping musical lines. By appropriately patching the equipment it is possible, for example, to program a bank of oscillators tuned in fixed relation to each other, as harmonic, subharmonic, or non-harmonic partials, with or without individual control of the envelope of each signal. To imagine the degree of variation possible, one need recall that all variables that are either frequencies or amplitudes[9] may be altered by manual shifts in baseline voltage, incremental voltages, or polarity, just as was the case with pitch.

Memory permits the tape reader to read each character separately, and store the information until, at a coded command, all the programmed characteristics of the musical event take shape simultaneously. Without memory the duration of a signal is limited by the number of successive repetitions of the specified code on the paper tape (and then, only in a reader that handles

[5] Programmed either by gating or by simple logic circuits.
[6] Frequency modulation refers to the periodic undulation of the frequency of a signal about a center frequency. The period is the duration of one such cycle and is inversely proportional to the frequency (usually low) of the modulating oscillator. The sweep width refers to the extent of frequency deviation from the center frequency of the modulated signal.
[7] Amplitude modulation refers to a periodic undulation of the amplitude about a mean amplitude. The period is the duration of one such cycle and is inversely proportional to the frequency of the modulating oscillator. The sweep width refers to the extent of amplitude deviation from the mean amplitude of the modulated signal.
[8] Ring modulation is a special type of amplitude modulation, in which the mean amplitude is zero and the amplitude of one input signal alternates in polarity at the rate of a second input frequency, resulting in an output consisting of the sum and difference frequencies of the two inputs.
[9] Frequencies: frequency, both variables of passband, period of FM and AM; amplitudes: amplitude, all components of envelope, sweep width of FM and AM.

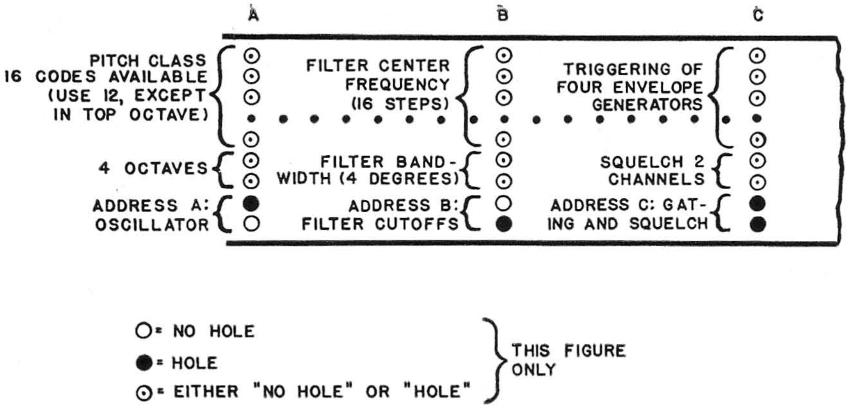


Fig. 5. Encoding scheme used for simple memory system.

successive holes as a continuous signal). If memory is to be included in the system, some special type of tape encoding method is necessary. To this end the designation "address" refers to the code identifying the specific pieces of equipment to which other encoded information is sent.

A simple memory system with three addresses may be used to describe the process. Fig. 5 illustrates the division of each character into those bits functioning as addresses and those bits bearing the information to be applied at the addresses. The information bits of addresses A and B are further subdivided. Character B, for example, is identified by the configuration of bits 7,8, where 7 is not punched and 8 is, as a voltage-controlled bandpass filter. In the same incremental fashion as was earlier described for pitch, bits 1-4 control the center frequency of the filter, and bits 5,6, its bandwidth. Address C is somewhat differently organized. Here bits 7,8, both punched, serve as the address for six gates controlling any combination of four envelope generators (patched through bits 1-4) and two "squelch" functions (patched through bits 5,6).[10] The actual envelope configuration is the sum of the manually controlled envelopes triggered at any one time. With more elaborate programming and memory, each of four independently variable parameters of the enevlope may be programmed. The output of each envelope generator may be patched to any of four loudspeakers or tape recorder channels.

Fig. 6 represents the encoding of three simple pitched events. Pitch is coded as in Fig. 4, and bandpass characteristics are coded in an analogous manner. Each tone, of sawtooth waveform, is modified by a bandpass filter whose center frequency rises in keeping with the pitch of the tone. The first and third notes begin with a wide passband which then narrows progressively at B′, B″, B‴. The resulting timbral changes will of course be heard only if the envelope is adjusted to be of sufficient duration at a given tempo. "Squelch" functions are coded at C′.

[10] The "squelch" functions are employed just prior to the encoding of the next event. Their purpose is to ensure complete decay of one tone prior to encoding the next.

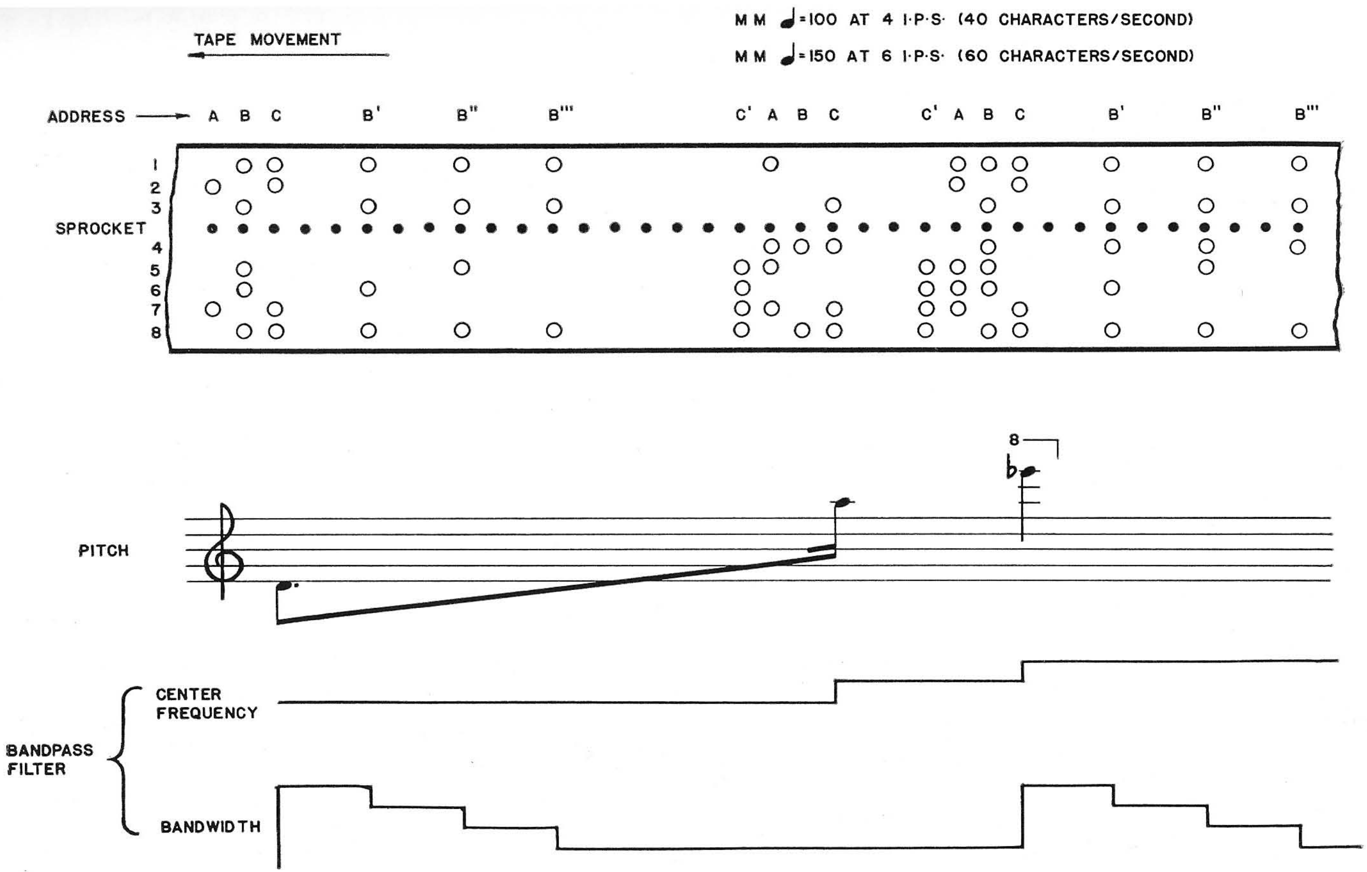


Fig. 6. Encoding of three simple pitched events.

The identical segment of perforated tape could also be used with quite different address designations. To illustrate: address A to bandpass filter with white noise input, and B to an oscillator generating sine tones; alternatively, code A may be used as an address of both an oscillator with sawtooth waveform and a bandpass filter adjusted to emphasize the fifth and sixth partials of all tones programmed. The output of the oscillator is now the input for the bandpass filter. B is now addressed to control frequency modulation (bits 1,2 for the frequency of the FM, bits 3,4 for its sweep), and amplitude (bits 5,6). Procedures of this type raise compositional questions as to the structural relevance of certain types of "automated variations."'

The musical example in Fig. 6 is intended only as a very simple illustration of a programming format. If one is to include a wide range of the variables previously listed, and in particular if independent sets of sound generating equipment are to be used for the synthesis of completely independent and overlapping lines of sound, a much more elaborate format is required. At a certain point, however, tape speed becomes a limiting factor. The more extensive the programming, the more urgent the demand for a faster tape reader.[11] This arises from the fact that the tape is being used for two conflicting functions, one analog, the other digital: (1) the precise reflection, in terms of distance, of the timing of musical events, and (2) the encoding of information relevant to the content of those events. As the space devoted to the digital coding assumes larger proportions, the only way to maintain timing accuracy and rhythmic flexibility is to increase the time-distance ratio so as to render the information space relatively insignificant. A corresponding increase in tape speed is then required in order to retain the desired tempo.

Central to this technique of electronic music synthesis is the preparation of the perforated tape. As mentioned earlier, simple programming may be accomplished manually by means of Port-a-Punch or mark-sensing cards, or through the use of a key-punch device. Vastly more elegant, and conservative of time and effort, is the application of a small computer to this end. In his article, Logemann discusses the various ways in which the computer may facilitate the synthesis of electronic music.[12]

The system described in this paper differs in many respects both from digital computer synthesis and from sequencer programming. It uniquely affords the capability of *both* programmed control wherein all events, although absolutely variable, retain some fixed and predictable relation to one another, *and* extensive manual influence in real time. Each musical event may be uniquely programmed with regard to a variety of parameters. Speeds up to 200 events per second are possible with simple character reading, and up to about 20 events per second with elaborate programming. As reel size is the only limit to the overall length of a segment of music, it is entirely

[11] or alternatively for the use of digitally coded magnetic tape where the bit density exceeds that of paper tape by an order of magnitude. This, however, touches on computer applications discussed in Logemann's article.

[12] In another contribution to appear in a subsequent issue of this journal, Logemann will present a symbolic notation for electronic music synthesis.

possible to include several thousand elaborately programmed notes or other events, or ten times that number of simply programmed events, per 500-ft roll of paper tape.

In contrast to the direct control of voltage-controlled equipment by a computer,[13] the intermediation of paper tape enables the composer to devote hours to compositional tryouts without tying up the computer. He is then free to more or less extensively modify, in real time, and by manual means, what he has programmed. By extending this process, compositionally relevant variations may be created. Just as in the course of writing instrumental music, new ideas suggest themselves during the process of imaginative (or realized) experimentation. From this point of view it may be desirable to start with relatively simple programmed control (for example, pitch, dynamics, and rhythmic relations). As the composer develops familiarity with the perceptual and musical counterparts of essentially electronic manipulations, he becomes increasingly able to make felicitous use of extensive programming.

---

[13] See James Gabura and Gustav Ciamaga's contribution to this symposium.

Symposium: Programmed Control

Techniques for Programmed Electronic Music Synthesis

George W. Logemann[1]

This paper deals with various methods for automatically controlling sound synthesizing equipment. In essence, these methods allow for changing interconnections or control voltages according to a programmed sequence. The methods complement techniques by which parameters may be varied in real time by the composer: a synthesizing instrument most desirably admits both automatic and spontaneous control.

Programmed control implies the existence of a medium through which the composer communicates his intent to the synthesizing devices. Ghent has proposed that paper tape be such a medium, the information regarding the specific interconnections and control voltages being punched onto the tape. To prepare his tapes, Ghent does not perforate the tape directly, but employs techniques whereby the information is punched on Port-a-Punch cards, and then punched on paper tape by a simple card-to-tape converter. Naturally, the work of preparing the cards can be left to a technician-assistant; the problem is that the details of pinpointing specific events require a fairly meticulous as well as reasonably clever assistant, the sort of person bored to insensibility by the tedium necessarily involved. This paper considers methods of automating the instructions normally given to the assistant—for example, interpretation of the notations the composer uses to express his intent. The first part of this paper deals with more powerful methods for preparing punched paper tape; the second with other media and methods which bypass the use of paper tape altogether.

Devices for preparing paper control tapes

One major problem in using paper control tapes is their preparation. Assuming an average control pulse density (i.e., the number of control characters per unit time) of ten per second, a five-minute piece requires 3000 characters; moreover, the characters usually must be punched at varying distances along the tape. Just the act of computing several thousand intervals and of punching several thousand characters by hand is a tedious and lengthy task which could be drastically reduced with some mechanical aid.

Typewriter devices which can punch paper tape exist, but usually provide only a limited subset of the $2^8 = 256$ available character combinations,

[1] I have enjoyed many conversations with Emmanuel Ghent (author of another contribution to this symposium) and gratefully acknowledge both his suggestions for additions to the present article and his introduction to the problems discussed herein.

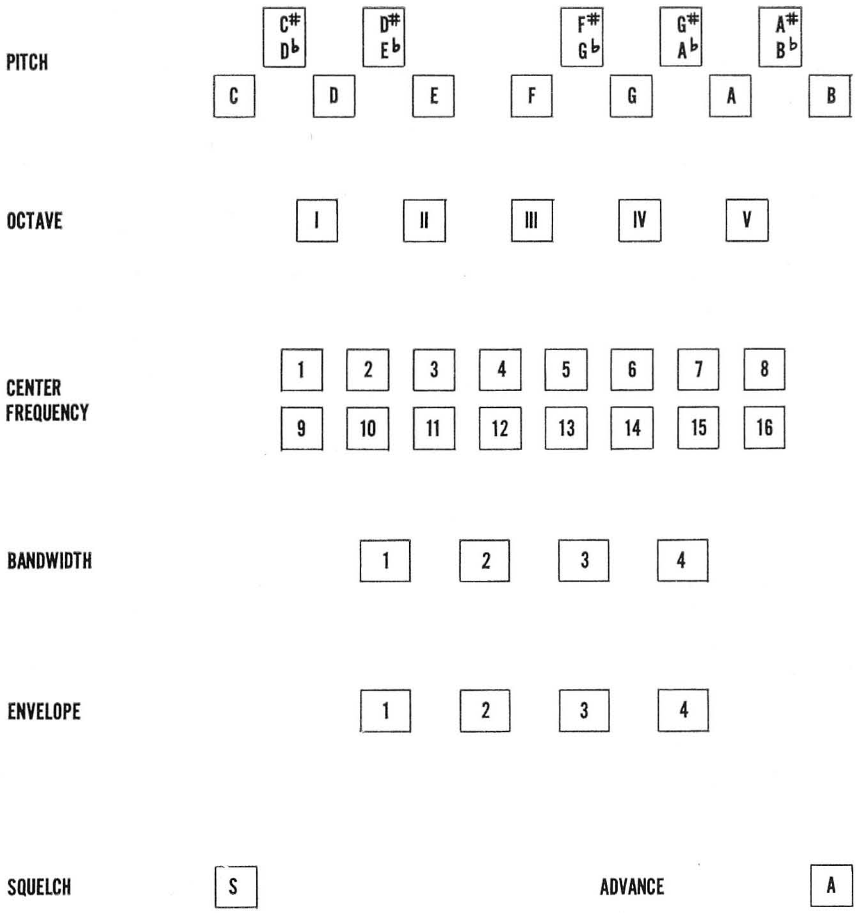


Fig. 1. A sample buttonboard

indicating that a careful choice of coding schemes must be made. In particular, no character can be demanded by the musical score which cannot be punched by some typewriter key. Even if such a coding scheme exists, there may be few logical relationships between the symbols on the typewriter keyboard and their musical effects, a source of error as well as inconvenience. Hence one must turn to mechanical devices especially intended for the purpose of creating musical control tapes.

The simplest device involves an array of nine push buttons, eight of which are wired to the eight punch magnets on a paper tape perforator, and the ninth wired to the mechanism that advances the perforator to punch the next character. Alternatively, one can wire a board with 256 buttons such that each button triggers the punching of one of the 2^8 characters and advances to the next character.

This technique is difficult to employ if the keyboard buttons are displayed in an order logically related to their respective lateral punch combinations; their display is not necessarily related to their use in the composition. A better method involves rewiring the buttons or placing the buttons on the board in such a way that the arrangement of buttons reflects their usage. For example, the combinations representing twelve semitone pitches can be connected to buttons arrayed in the manner of a piano keyboard; the register buttons can also be grouped together. The buttonboard can be permanently linked to the punch magnets on the paper tape perforator, or the connections can be patched semi-permanently for current composition. Fig. 1 represents a buttonboard array for punching the three-address encoding scheme incorporating memory proposed by Ghent. The pitch and center frequency buttons would punch their respective address codes as well as their respective binary four-channel character, as would the envelope and squelch buttons. The octave and bandwidth buttons would advance the paper tape after punching their appropriate character. Thus each control character requires depressing no more than two buttons, and the buttons are significantly positioned.

Similar considerations are involved in preparing punchcards to be converted to paper tape. Either the composer must punch individual holes by hand, e.g., with a single-column punch or directly onto Port-a-Punch cards, or he must employ a standard keypunch whose character set may not cover the required set of punch column combinations. If one wishes to avoid multiple punching, he must wire a buttonboard into the keypunch magnets, a somewhat more difficult procedure than similarly wiring a paper tape perforator.

Paper control tapes employ longitudinal distance to encode duration. Although the analogy between time and distance is conceptually useful and valid for visually locating and comparing events on the paper tape, it presents a considerable problem in tape preparation. The problem arises in reckoning the number of blank characters to punch along the tape in order to achieve the desired time lapse between events. From the time lapse between consecutive events, one must compute the tape spacing between consecutive control groups as well as the number of characters of control information.

Unequal time lapses, or lapses generated by superimposed rhythms, present greater difficulties than events spaced at regular intervals. In the simpler case of regular intervals, one could employ a stepping relay to facilitate punching the required number of blank characters. The relay would be initially set to a number equal to the spacing of fundamental beats. As octave, bandwidth, envelope, and advance buttons are depressed, the relay is stepped down; finally, to reach the next event, a runout button is touched whereupon the remaining advances are triggered by the relay stepping down to zero; the relay then resets to the desired spacing number. A refinement would allow buttons or keys selecting various spacing numbers. To handle cases of regular polyrhythms—for example, a sequence consisting of a series of equally spaced events that fit between the equally spaced events of another series, such that no superimpositions occur—one can run the paper tape through the punching process two (or more) times. To handle superimpositions, one could construct a system whereby a second tape is

superimposed upon a master in such a way that a character from the second tape coinciding with a character on the master is punched only in the first blank position in the master.

Similar problems arise in preparing punchcards, although the drum card on the keypunch can be used to set up tab stops corresponding to any fundamental beat which evenly divides 80 (the number of columns on the card); other multiples can be utilized if the card-to-tape converter can be programmed to ignore the superfluous columns at the end of the card. Alternatively, one can develop a system with column counts independent of natural card boundaries; in this case one can expect no tabulating help from the keypunch.

Preparing control tapes with a computer

In controlling a number of channels in parallel, a new problem arises. Namely, events which occur simultaneously must be directed with a succession of punched characters if a single control tape is used, yielding a minute discrepancy between the starting times of desirably simultaneous events. Conceivably, electronic delays can postpone the response of the first channel to be given control information so that it reacts in parallel with the second channel, etc. One solution discussed by Ghent involves parameters set up by his addresses A and B, with the sounds triggered and terminated by address C. Or, faster tape speeds can be employed so that the discrepancy becomes unnoticeable. Faster tape speeds are also desirable to alleviate the error introduced by coding durations that do not correspond to an even multiple of the time increment corresponding to adjacent characters punched on tape: the faster the tape, the smaller the "round-off" error.

Since mechanical paper tape readers have a practical maximum reading speed of several hundred characters per second, other devices must be brought in. Photoelectric readers can run at kilocharacter rates; variable-speed, reversible photoreaders are feasible. Magnetic tapes used with digital computers might be the medium to supplement paper tapes: since the recording density of digital magnetic tapes is several hundred characters per inch, any desired character frequency transfer can be achieved with little difficulty. There are devices by which one can write characters directly onto magnetic tape, much as one would directly punch paper tape. Here, too, variable-speed, reversible digital magnetic tape drives are available, the only serious drawback being a cost greater by orders of magnitude. Furthermore, digital magnetic tapes are not easy to read visually, as are punched paper tapes, necessitating monitor and editing equipment. In either case, with an increased number of control characters required per second, preparing tapes by hand becomes enormously more difficult. Thus one must turn to computer techniques to produce either perforated or magnetic tapes.

Many contemporary computers have typewriters of one sort or another connected directly to the memory of the machine, and many provide for buttonboards that one can easily program to have the logical display described in context with Fig. 1. The problem of rewiring such a buttonboard now becomes relatively trivial, since only the pulse-analyzing computer program

Table 1. Rules for writing a simple symbolic code

1. The format for encoding a control tape is: *code/code/code/code/ . . .*
2. If the *code* is a number, that number of blank characters is to be punched. (No code at all, i.e., //, represents *one* blank. Note that /// is equivalent to /2/.)
3. Otherwise, the *code* represents precisely one character, directed to precisely one of the three addresses as follows:

   to A: *pitch,* one of the letters *A* through *G* preceded possibly by *X* for sharp or *Y* for flat;
   octave, the letter Φ preceded by a numeral *1* to *5* indicating the desired octave;

   to B: *center frequency, nQ,* where *n* is a numeral *1* to *16* stating which of 16 center frequencies is selected;
   bandwidth, mW, where *m* is a numeral *1* to *4;*

   to C: *envelope(s),* one to four of the numerals *1, 2, 3,* and/or *4* (in order) followed by letter *N;*
   squelch, simply the letter *S.*
4. All items of the symbolic code of a particular character must be supplied.
5. Blanks are ignored and may be used to facilitate legibility.

Table 2. A sample of symbolic code

D2Φ/5Q4W/12N/2/5Q3W/2/5Q2W/2/5Q1W/5/S/A3Φ/8Q1W/
34N/2/S/YE5Φ/13Q4W/12N/2/13Q3W/2/13Q2W/2/13Q1W/5/.

need be changed. For output, the on-line typewriter-tape perforator system may be inadequate both in speed and character set; however, auxiliary independent perforators are available (naturally, at additional cost).

However, the greatest possibilities lie in the direction of using the typewriter as the input mechanism (or, equivalently, cards punched with free-field alphanumeric information). The problem is to design an alphabetic language related mnemonically and logically to the functions one wishes to control. In addition, one can provide a symbolic notation for the time lapses between control events, e.g., the number of character positions to be spaced. Once one has prepared the symbolic language, one can write a computer program to translate elements of the language into any desired pattern of punched or magnetic tape characters.

For example, one can define a language for the three-address system referred to by Ghent and Fig. 1, using Table 1. In general, the symbolic code for punching tape is a string of letters and numerals separated by /'s. Between two /'s, one states either the code for a single character or a number indicating that number of spaces (blank characters). For example, Table 2 gives the symbolic code for the three-note example of Ghent.

To translate a string of characters such as one generated by the rules stated above, one might construct a translation table giving all possible address codes and their respective translation characters to be punched on paper tape. A computer program to isolate the symbolic code and to search the translation table for the code thus discovered is fairly simple to write, but

Table 3. A translation table for the symbolic code elements

Substring	Translation	Substring	Translation
C	00000001	Q	00000010
XC	10000001	1Q	00000010
YD	10000001	2Q	10000010
D	01000001	3Q	01000010
XD	11000001	4Q	11000010
YE	11000001	5Q	00100010
E	00100001	6Q	10100010
XE	10100001	7Q	01100010
YF	00100001	8Q	11100010
F	10100001	9Q	00010010
XF	01100001	10Q	10010010
YG	01100001	11Q	01010010
G	11100001	12Q	11010010
XG	00010001	13Q	00110010
YA	00010001	14Q	10110010
A	10010001	15Q	01110010
XA	01010001	16Q	11110010
YB	01010001		
B	11010001	N	11110011
XB	00110001	1N	10000011
YC	11010001	2N	01000011
		3N	00100011
		4N	00010011
Φ	00000001	12N	11000011
1Φ	00000001	13N	10100011
2Φ	00001001	14N	10010011
3Φ	00000101	23N	01100011
4Φ	00001101	24N	01010011
5Φ	00111101	34N	00110011
		123N	11100011
		124N	11010011
W	00000010	134N	10110011
1W	00000010	234N	01110011
2W	00001010	1234N	11110011
3W	00000110		
4W	00001110	S	00001111

Note: Φ, *W*, and Q are set by default to *1*Φ, *1W*, and *1Q*, respectively; *N* alone becomes *1234N*. Also, octave *5*Φ is only an extension of *4*Φ, and can only be used with notes *C* through *YE*.

the translation table becomes cumbersomely long. Rather, one can simplify the table by giving only the permissible substrings of codes between two /'s; in particular, one can list all possible substrings that end in one of the letters *A* to *G*, *N*, Φ,[2] Q, *S*, *W*, here called the *action* letters. Action letters may be preceded (qualified) by *passive* symbols, here the letters *X* and *Y*, and all numerals. Table 3 gives all possible occurrences of substrings containing exactly one action, always the letter in rightmost position. Note that unqualified action letters can be given a qualification by default; normally *N*, Φ, Q,

[2] The letter O is written Φ to avoid any confusion with the numeral 0.

and *W* must be preceded by numerals.

To make use of the table, one needs a computer program that will superimpose the character patterns for all action letter substrings found between a particular pair of /'s, punching the character only when the second / is read. If no action letter is found, the string between two /'s represents a number of blank characters to be punched; adjacent /'s separated by no symbols whatsoever indicate *one* blank (by default). The appendix contains a PL/I program describing the algorithm in detail.

Naturally, one could write a fairly sophisticated code geared to translating the particular choice of notation used as an example; however, the present system allows arbitrarily free changes in symbols. In particular, one might choose four distinct single symbols to represent envelope patterns *1N, 2N, 3N,* and *4N,* respectively, giving a more compact as well as more meaningful notation. A useful refinement would be to build in the notion of redundancy, so that only those substrings that change need be punched. For example, in Table 2 the center frequency of the filtering remains constant while the bandwidth changes. We need therefore write only . . . */5Q3W/2/2W/2/1W/* . . . if the computer program has been modified to "save" the symbols *5Q.*

Moreover, one can use the techniques of translating a symbolic code to admit a greater notational freedom than one might otherwise achieve with a simple one-for-one alphabetic representation of tape characters. For example, rather than use an absolute pitch notation, one might employ a baseline-deviation scheme, in which pitches are always stated in terms of interval above or below a fundamental pitch, which itself varies throughout the composition but is stated in an absolute notation. Therefore, it is not necessary to devote two separate character fields to pitch control; the computer can perform all conversions to absolute pitch, necessitating only one pitch field. In the above example, the pitch and octave references, while *stated* distinctly, could be programmed as one six-channel field, admitting a range of 64 notes as opposed to 50 obtained using the less efficient character coding scheme of the example.

Finally, instead of representing time lapse between control characters explicitly as a number of character positions to be spaced, one can represent longitudinal tape distance in terms of real seconds or in terms of a fictitious duration notation such as "beats." A time lapse stated in seconds, multiplied by the intended tape speed in characters per second, yields the corresponding number of character positions; more generally, from units in some arbitrary beat system one can obtain real seconds using, e.g., metronome markings. To obtain the number of spaces one subtracts from the desired character spacing the number of characters of control information, all work being done by the computer program. With a little effort, the computer program can be made to place a number of events between occurrences of two others according to some simple law, e.g., equal spacing.

Direct computer control

The methods for computerized tape preparation discussed in the previous

section can also be used to obviate the need for such tapes. All that is required is an eight-channel line running directly from the computer to the decoding device; now it is essential that the score be communicated to the computer in a symbolic language.

Several questions arise if one is controlling the synthesizing apparatus directly from the computer. One question is whether the computer can translate the symbolic notation into control pulses in real time—an unsophisticated program such as that described in the previous section may be too slow. Perhaps the computer will need to translate the symbolic notation to control pulses stored completely within its memory and called out only when played; even though the translating cannot go on in real time, the playback process can. Another serious difficulty arises if one wishes to control the speed and direction of execution in the manner one could control tape movement with a mechanical reader. Effectively, one needs techniques for real time *input* to the computer, of parameters which can vary spontaneously. The existing techniques fall into two general classes, depending on whether the input information is basically digital or analog in form, i.e., in discrete quantities or continuously variable.

Digital input devices include typewriters and buttonboards mentioned earlier. One can program to reset translation parameters (such as the metronome interpretation of the fictitious time in terms of real time units) according to information read from the buttonboard or keyboard. Or, more generally, one can direct with one button that the parameter start increasing (at a pre-assigned rate), touching another button to stabilize the parameter at a new value. Other buttons can be used to indicate that the symbolic coding scheme is to change, e.g., that information sent previously to one channel is now to be sent to all.

Analog devices provide a directly variable input; the simplest example is the potentiometer, which allows a parameter to vary continuously at the twist of a knob. We might call such devices "one-dimensional" inputs, in contrast to "two-dimensional" input devices in which a rotating sphere or lever with two degrees of freedom can control two parameters independently. For example, the forward-back direction might be interpreted as center frequency, with the left-right axis controlling bandwidth. A battery of one- and two-dimensional controls offers great flexibility.

Finally, one should consider graphical communication devices, the most familiar of which are the light-pen/cathode-ray tube combinations. Actually, the computer can draw curves and figures (as well as symbolic characters) on the face of the tube; if the curve is drawn following the tip of the light-pen, one has the effect that the pen is actually drawing the line. Thus the composer can draw continuous graphs and curves indicating the extent and shape of filter functions, envelopes, and melodic contours. One can conceive of the composer drawing a diagram of (rather than physically wiring) the connections of his apparatus; moreover, rewiring could be done as the composition is being played. In these examples the graphical device behaves as a continuous input mechanism.

Graphical devices also admit discrete inputs; in effect, the cathode-ray tube draws a buttonboard; the light-pen touching a "button" can be said to "depress" the button. Since the button display can be varied instantaneously, the display tube can present many more alternatives than any physical board. Thus the composer can have virtually all the parameters of his instruments literally at his fingertips, at all times. Nevertheless, the changes the composer realizes in such an improvisatory manner can be retained for future analysis and reworking.

The topics discussed in this paper are fundamentally introductions to the vast spectrum of mechanisms and techniques available today. The examples are quite simple; further ramifications of the technique of symbolic input are left to a future paper.

Appendix: *A computer program to translate symbolic code into characters*

```
TAPEOUT: Procedure (PASSIVE, ACTION, IN, OUT, INFILE, OUTFILE);
  Declare /*parameters*/
    PASSIVE character(*) /*passive input symbols*/,
    ACTION character(*) /*action input symbols*/,
    IN(*) character(*) /*recognized input strings, lefthand column of the
        translation table*/,
    OUT(*) bit(8) /*character for output for recognized input, from right-
        hand column of translation table*/,
    INFILE file /*input symbol stream, containing code*/,
    OUTFILE file /*output characters, to be punched*/;
  Declare /*local variables*/
    SYMBOL character(1) /*current symbol being studied*/,
    STRING character(20) varying /*current string to be translated*/,
    TRANSLATION bit(8) /*translation of current string*/,
    COUNT decimal fixed initial(0) /*count of punched characters*/;
  On endfile (INFILE) go to FINISH;
  On error go to DONE;
  On conversion Begin; Put skip list ('Non-numerical passive character',
    ONCHAR, 'within // discovered after output character', COUNT);
    ONCHAR = '0'; End;
SETUP: STRING = ''; TRANSLATION = '00000000'B;
OBTAIN: Get file (INFILE) edit (SYMBOL) (A(1));
  If SYMBOL = '' then go to OBTAIN;
  Else if index (PASSIVE, SYMBOL) > 0 then do; STRING = STRING || SYM-
      BOL; go to OBTAIN; End;
  Else if index (ACTION, SYMBOL) > 0 then do; STRING = STRING || SYM-
      BOL;
    Do I = 1 to hbound (IN, 1);
      If STRING = IN(I) then do;
     TRANSLATION = TRANSLATION | OUT(1); go to OBTAIN; End;
      Else End;
```

```
        Put skip list ('Error —', STRING, 'not found in input list; blank inserted
            after output character', COUNT);
      Call WRITE ('00000000'B);
      End;
   Else if S = '/' then do;
      If TRANSLATION then call WRITE (TRANSLATION);
      Else if STRING = '' then STRING = '1';
      Do I = 1 to STRING; call WRITE ('00000000'B); End;
      End;
   Else put skip list ('Illegal character', SYMBOL, 'discovered after output char-
        acter', COUNT, 'and ignored');
Go to SETUP;
WRITE: Procedure (CHARACTERBITS);
   Declare CHARACTERBITS bit (8), PUNCHCHARACTER character (1);
   Unspec (PUNCHCHARACTER) = CHARACTERBITS;
   Put file (OUTFILE) edit (PUNCHCHARACTER) (A(1));
   COUNT = COUNT + 1;
   Return;
   End WRITE:
FINISH: Put skip list ('Output file completed with', COUNT, 'characters');
DONE: Close file (INFILE); Close file (OUTFILE); Close file (SYSPRINT);
Return;
End TAPEOUT;
```

Explanation

Basically one scans the input string of symbolic code, written on file INFILE, to find a substring STRING ending on one of the action characters; upon finding a complete STRING one scans table IN, finding STRING in position IN(I). Then one superimposes the corresponding translation OUT(I) upon the pattern TRANSLATION.

Upon reading the character / punches TRANSLATION (by writing onto file OUTFILE) if indeed some character code has been specified; otherwise one punches the specified number of blank characters.

In particular, the action characters all occur in the string ACTION; the index of SYMBOL (the current input symbol being scanned) in ACTION is greater than zero if and only if SYMBOL is found in ACTION. Passive symbols are collected together into STRING until an action symbol is encountered.

All punching is done in subprocedure WRITE. Error messages are printed out if an illegal character is encountered violating the syntax of the language.

Symposium: Programmed Control

Digital Computer Control of Sound Generating Apparatus for the Production of Electronic Music

James Gabura and Gustav Ciamaga

It must be apparent to observers of present day electronic music that many composers continue to express themselves effectively with the conventional studio techniques of the 1950's. Electronic music is closely linked with its most important tool, the tape recorder, and the "classical" techniques of tape manipulation as developed in the '50's are basic to the electronic composer's craft. It is equally apparent that there are some composers who, dissatisfied with conventional working procedures, are turning to programmed control methods for realizing electronic music. The potential for more efficient techniques in the electronic medium has existed from its early beginnings but their extensive application has been retarded by insufficient research, lack of suitable apparatus, and often by prohibitive costs. The present paper offers, to composers having access to computing facilities, a simple method for using a computer as an aid in the production of electronic music.

All methods for automating the production of electronic music have in common the principle of coding or programming various parameters of the electronic sound material. At a simple level, this programming might involve setting a function generator to generate simple melodic patterns from a voltage controlled oscillator; or at a more sophisticated level, punching a paper tape which in conjunction with a suitable reader activates auxiliary equipment for controlling pitch, time, amplitude, and timbre.[1] But possibly the most important tool for an automated approach to electronic music is the modern hybrid digital-analog computer.

Some excellent programs using digital sampling techniques have been devised for generating sound, notably the Bell Telephone Laboratory MUSIC IV and the Argonne Laboratory MAESTRO programs. Though less publicized, the program developed by Ercolino Ferretti of M.I.T. has displayed enormous potential. A method that features a combination of digital and analog techniques has been devised at the Coordinated Science Laboratory of the

[1] See Emmanuel Ghent's contribution to this symposium.

University of Illinois.

Another approach which utilizes a combination of digital and analog techniques is presently under exploration at the University of Toronto. With this method, the computer does not generate the actual waveforms, but rather it provides voltages for controlling pitch and amplitude patterns of existing oscillators and level control amplifiers.

The utilization of the computer in this manner has obvious advantages:
a) The nature of the equipment is relatively simple, i.e., the computer can be smaller.
b) The auxiliary equipment is available at most studios.
c) The controlling programs are simple to write and even simpler to use.
d) The end product can be monitored in real time, allowing immediate modification of tempo, pitch, range, timbre, etc.
In the initial experiments the following basic equipment was used:
2 R.A. Moog voltage controlled oscillators
2 Canadian National Research Council level control amplifiers
1 IBM 1710 digital control system
1 IBM 1711 digital-analog interface

The voltage controlled oscillators have a large pitch range and generate sawtooth, square, variable-width pulse, and triangular waveforms; the level control amplifiers are capable of wide range dynamic control with low distortion. With this equipment many examples of two part textures were realized with some success; with additional oscillators and amplifiers, a five part texture can be obtained without resorting to multiple recording and dubbing techniques.

The special hardware essential for this technique is the digital-analog interface that makes available various analog functions in the form of computer controlled potentiometers, relays, and latching contacts. For this application only the potentiometers are used. Each potentiometer (ten are available at the interface) can be set by the computer to any of 999 positions. In practice, a fixed voltage supply is placed in parallel with each potentiometer used. For each voice, the wiper on one potentiometer presents a programmed control voltage to the input of an oscillator. The output of the oscillator is presented in turn to a level control amplifier which responds simultaneously to a programmed control voltage from a second potentiometer to release the sound with its appropriate duration and envelope.[2] For a two part texture, four potentiometers are required—two for the oscillators and two for the associated level control amplifiers (see Fig. 1). If desired, other control voltages could be presented simultaneously to voltage-controlled filters to alter the harmonic content of the basic waveforms offered by the oscillators. Alternatively, a formant filter (or other modifier) could be inserted after the oscillator and before the level control amplifier to achieve similar results.

To maintain accurate control of the sound parameters a simple calibration

[2] Since the control voltages consist of rectangular steps, a simple smoothing filter is required before the control input of the level control amplifiers.

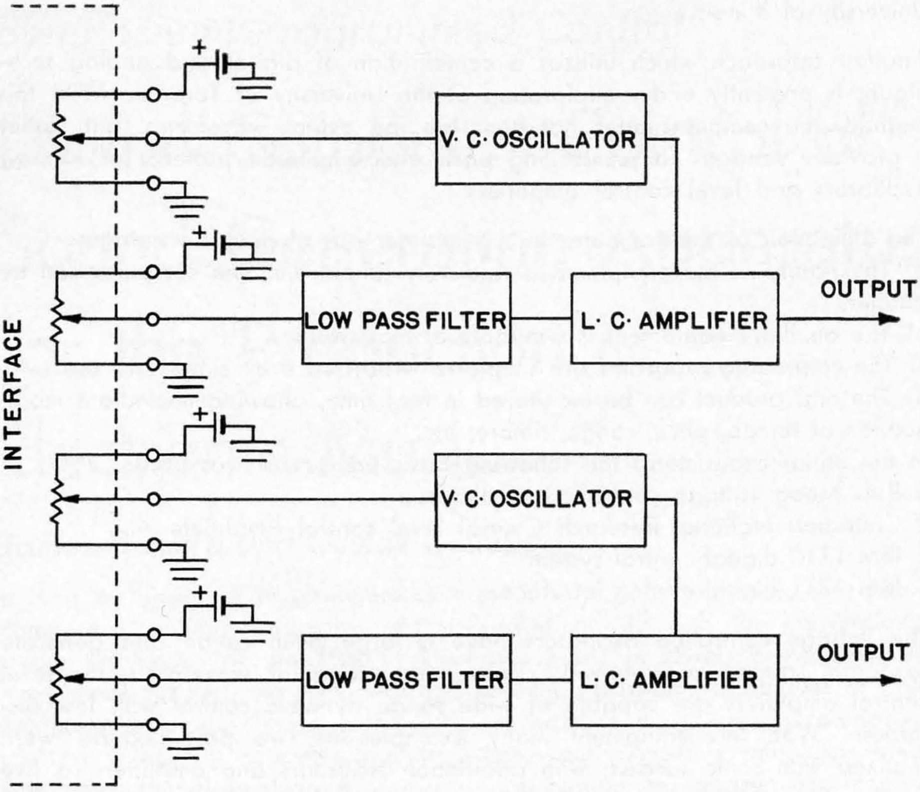


Fig. 1. Basic Equipment Configuration

procedure is necessary. With the control voltage supply connected, the computer sets the potentiometer to each of 20 equally spaced positions, while the output of the associated equipment is measured.[3] This data is then punched on cards and the intermediate values are interpolated by the computer. The running speed is calibrated by timing a sample run with a stopwatch.

The digital computer program is written in FORTRAN and utilizes locally defined subroutines for controlling the output potentiometers. The coding for the music is simple and can be performed quickly with a little practice. The composer must specify:

a) the duration of the basic rhythmic unit in milliseconds.

b) the envelope of the particular "instrument" or part. The envelope definitions are similar to those used by the MAESTRO program.[4]

c) the frequency and depth of both tremolo and vibrato. Random fluctuations in these parameters can be introduced if desired.

d) for each note, a series of four numbers denoting pitch class, octave, duration, and peak amplitude.

---

[3] The entire calibration procedure could be automated using an analog-digital converter in conjunction with suitable measuring equipment.

[4] Robert K. Clark, *MAESTRO, A Program for Real-Time Generation of Musical Sounds,* Audio Engineering Society Preprint no. 428, from 17th annual meeting, Oct. 11-15, 1965.

This paper has outlined briefly one method for digital computer control of sound generating apparatus. Even with its inherent limitations, the program provides the interested composer with a flexible and economical approach for automating some aspects of electronic music.

"Introduction to the field of happenings by its great master . . ."
Book News

". . . The most concise and clear statement of the thinking behind these experiments. . ." *Art and Artists*

A Primer of Happenings & Time / Space Art

by Al Hansen

160 pages, with over 100 illustrations: $4.50

". . . Not a cool dissertation on art criteria but a rather fascinating picture. . ." Udo Kasemets in *The Literary Times*

"*Postface* is the critical and reflective part. . . *Jefferson's Birthday* is a series of explosive recipes for performance. . ."
Juliet Arning Siragusa in *Collage*

Jefferson's Birthday / Postface

by Dick Higgins

384 pages, with 115 illustrations: $5.95

SOMETHING ELSE PRESS, INC.
160 Fifth Avenue, Room 615, New York, N.Y. 10010

Remarks to the Kind Lady of Baltimore

delivered at a concert of his works at Hunter College, New York, on December 1, 1965, by

Luciano Berio

Not long ago, after a lecture in Baltimore, during the ritual of the "question period" that every traveling salesman of contemporary music has to deal with, a very kind lady asked me, "Mr. Berio, how do you relate your work to life?" Because of my lifelong habit of over-reading and generalizing, I assumed that by "life" she meant not only my personal life, but also the life of mammals in general, myths and religions, nuclear war, the historical development of the arts, neurosis, love, and astrology. Practically strangled by that question, I was only able to answer that I didn't know precisely and that if I thought I knew maybe I wouldn't even write music, but books. I said that in any case a complex tissue of relations exists and that whatever we do—not necessarily music—is an attempt to uncover a part of it and to become more aware of what we are and where we are. I also said to her that I feel there is always something untrue about a composer talking about himself and that to me the most illuminating self-portraits are those in which a composer doesn't speak about himself but about others or about something other than his work, as were the cases of Debussy writing about the paintings of Turner or the music of Moussorgsky, Webern writing about Schoenberg, and Stravinsky about many others. Maybe the kind lady of Baltimore reached the conclusion that, for me, there is very little about music that can be directly verbalized (unless one is dealing with specific problems in a classroom or of a performance, or is content with metaphors) and that, for me, music must be something like a mystery game with crowds of composers and concert-goers wandering about, searching for the ultimate meaning of their actions.

Because I have agreed to say a few words before tonight's performance, let me use this occasion to answer the kind lady of Baltimore more properly, even if she is not here to hear my answer. My answer is this:

Once, in Baltimore, there was a lady who decided to go to hear a lecture-demonstration on "Language and Music" given by Luciano Berio. She went because she was hoping to learn something more about the relation between music and the life of mammals, the society in which she was living, racial problems, love, and astrology. At the end of the lecture she seemed very stimulated and she asked the composer what, according to him, was the relation between writing music and her going to listen to it. The composer said that finding an answer to that question is inherent to the role of every individual member of an audience and that *she* should be questioned and

she should answer, not he. He also said that writing music and going to listen to it is perhaps like a mystery game full of crowds searching for the ultimate meaning of their actions and thoughts. But the composer felt that this, again, was not the proper answer and—a few weeks later—he felt the need of giving the kind lady of Baltimore a better answer, even if she was not there to listen to it. And the answer was:

Once, in Baltimore, an immense crowd (at least fifty times larger than tonight's audience) went out to educational institutions, to theatres, to concert halls, to libraries, to churches, and to courthouses, searching for the ultimate meaning of their actions and thoughts. It was night and they couldn't find very much: only an Italian composer who was talking about vocal music and explaining something at the blackboard. After the lecture a kind lady asked the composer how it would be possible to uncover at least a small part of the complex tissue of relations that undoubtedly exists between music and any experience outside of it. Dealing with such a loaded subject, the "question period" soon became an overheated discussion about sound and meaning, about "words and things," about the duality of language, about transposition, translation, and metaphors, about grammar and poetics. There was no moderator to calm the spirits down and very soon, according to the dynamics of group interaction, people were facing each other, separated into two different fronts. The discussion degenerated very soon into a ferocious fight; the kind lady disappeared in the turbulent crowd and the composer stood in a corner to watch. It was almost beautiful and everybody seemed to be right, although the issue appeared to be mainly a disagreement about terms, about the labels of things rather than the things themselves, about the impossibility of labeling certain things and the significance of those things that cannot be labeled. However, watching that confusion, it seemed that the two groups could be quite easily labeled as "operationalist" on one side and "structuralist" on the other. Very soon the former group was yelling about twelve-tone sets, note-objects, combinatorial procedures, and pitch coherence. The latter, the "structuralists," about meaning, segmentations of the sound continuum, synchronic and dyachronic views, history, and responsibility. Gradually the crowd turned into a mob and finally only a few words could be grasped here and there: chance and determinism, good and evil, East and West, black and white, body and soul, night and day, mommy and daddy . . .

That night never ended and here I am again, facing once more the impossibility of giving a real answer to the kind lady of Baltimore. What can we do? Because I have agreed to say a few words before tonight's performance, let me use this occasion to say good-bye to her, even if she is not here to hear me, and to apologize.

Contributors

Luciano Berio is currently teaching at Juilliard School of Music.

Harald Bode is Chief Engineer at Bell Aerosystems, Buffalo.

Robert Ceely maintains a personal electronic music studio in Boston.

Gustav Ciamaga is director of the University of Toronto Electronic Music Studio.

James Gabura is a Ph.D. candidate at the Computer Science department of the University of Toronto.

Emmanuel Ghent composes at the Columbia-Princeton Electronic Music Center.

George W. *Logemann* is Coordinator for Computer Sciences at the Institute for Computer Research in the Humanities, New York University.

Robert A. Moog is Technical Editor of EMR and president of the R. A. Moog Co.

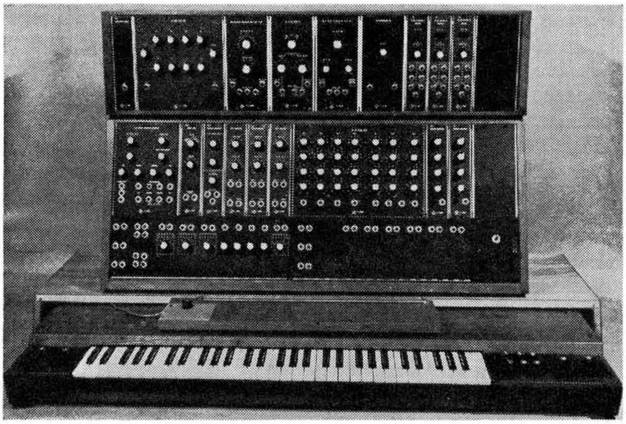
Karlheinz Stockhausen is visiting lecturer at the University of California, Davis.

Raymond Wilding-White is director of the Case Institute of Technology Studio for Experimental Music.

Translator *William Sylvester* is Professor of English at the State University of New York at Buffalo.

The Moog Synthesizers

Integrated standard consoles for electronic music composition and performance



The Synthesizer II

First to introduce a complete line of modular instruments for audio signal generation and processing, the R. A. Moog Co. now offers three standard, integrated systems. Designed for both electronic music studio composition and concert performance, the Synthesizers are capable of a complete range of periodic, transient, and random signal generation, and offer comprehensive filtering, modulation, reverberation, synchronizing, and mixing functions. Their simple, versatile operation and conservatively-rated solid-state circuitry make the Moog Synthesizers uniquely appropriate for use wherever flexible, accurate signal generation and processing is required.

The new Moog Catalog contains detailed information and specifications for these and other products.

Address your request to Dept. E.

R. A. MOOG Co.
Trumansburg, N. Y.
14886

See our exhibits at the Music Teachers National Association Convention, Sheraton-Jefferson Hotel, St. Louis, April 18-21, and at the Audio Engineering Society Convention, Hollywood-Roosevelt Hotel, Los Angeles, April 24-27.

Electronic Music Review